THE ROTEL HOME THEATER and HI FI ENCYCLOPEDIA

ROTEL

DEDICATION & ACKNOWLEDGEMENTS

This book is dedicated to Jack Anthony, an independent sale representative *par excellence*, whose recent death after a long battle with cancer saddened many in the industry he helped to build. A mentor, and, most importantly, a friend, Jack gave me far more time and guidance than common sense might have dictated. In so doing, he taught me the real meaning of the word "professional."

Books don't magically appear in a vacuum. At least this one didn't. A number of people contributed.

First, thanks to Scott Rundle of Rotel for his consistent patience and persistent cajoling. His mandate that this book be as objective and complete as possible merits high praise. That touch-of-class attitude isn't often evident in today's "eat your competitor" environment.

Hyam Sosnow also deserves mention. Many years ago, he provided the framework for a related book that never saw the day. Well, Hyam, here's some sunlight!

Roger Dressler and Brent Butterworth read and commented on all matters Dolby. Likewise, John Dahl of THX, Lorr Kramer of DTS, and Mike Ritter of Microsoft's HDCD team reviewed material of particular interest to them. Much appreciation to all.

On a less personal level, I'd like to thank two people I've never met or spoken with: Jim Taylor and F. Alton Everest. Jim's book, *Demystifying DVD*, is *the* source for technical information on this exciting technology. Likewise, Everest's *The Master Handbook of Acoustics* is an indispensable source for anyone trying to understand how rooms and speakers influence each other. I recommend both volumes enthusiastically.

But the last and most important acknowledgement is to everyone who will read this – or parts of it – to learn something you might not otherwise know. Thank you for your trust and effort.

Len Schneider
TechniCom Corporation
January, 2003

AN OPEN LETTER TO OUR CUSTOMERS

Balanced Design Concept.

I will always remember my first meeting with Bob Tachikawa, Rotel's President and son of the company's founder. It was in Paris at the 1980 Festival du Son (Festival of Sound) audio show, nearly a year after I joined Rotel's subsidiary in the U.K. Over dinner that night, our discussions led to the fact that many audio manufacturers were then producing highly "featured" models. Since Bob and I both shared a passion for music and genuinely abhorred listening to mediocre hi-fi, we soon fell into a determined mind-set to buck the trend and try to carve out a strong niche for Rotel by other means.

We formulated a unique idea — a slim, high performance amplifier to be designed by a local British engineer. As we had already produced high-powered, critically recognized amplifiers in our own factories, we knew that credibility was not a problem. Our greatest challenge was engineering a high level of performance at an affordable price.

That first integrated amplifier, the RA-820B, was elegantly simple and formed the basis of our Balanced Design philosophy: refined implementation of sound engineering principles. The goal, then as now, was to achieve the highest performance by using only quality components. And if that meant a Rotel product might not have some barely used features, so be it!

The approach was, pardon the expression, a sound one as the RA-820B became a local legend and went on to win numerous awards in the U.K. It made Rotel a household name among those who knew fine audio.

Today, we follow these same roots. Rotel's success, we believe, lies in our ability to consistently produce electronics that both fulfill our customers' needs while sounding as if they cost far more than they do! That's why we've written this book. We think it will help you better understand both the technical features and the philosophy behind every Rotel product and we invite you to join us as we explore the true value of Rotel.

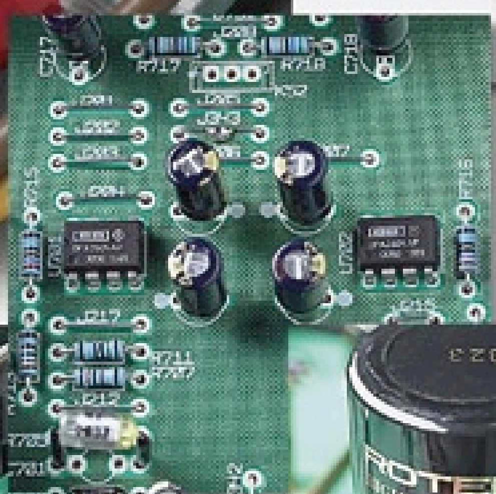
More importantly, please take some time to gather a selection of your favorite music or movie soundtracks. Turn on a Rotel system and listen. We think our commitment to better sound will speak for itself.

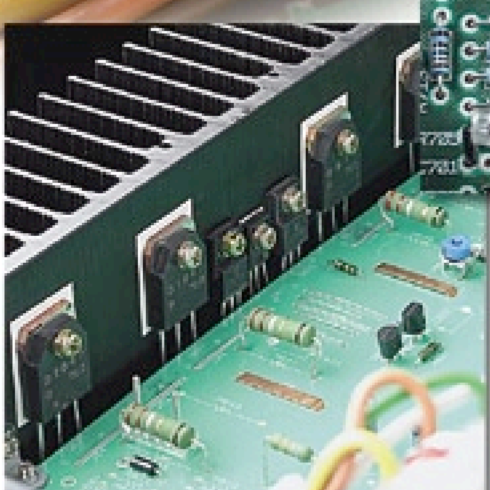
Thank you.

Sincerely,

Michael Bartlett
Vice-President and General Manager
Rotel of America

Balanced Design Concept.





First, our thanks! The first edition (*A Guide to Better Hi-Fi and Home Theater*) of this book was so warmly received by our dealers that we couldn't always supply copies when people asked for them. The enthusiasm was, in a word, overwhelming and it showed us that an objective book about audio and home theater was really needed.

In the six years since that book appeared, much has changed. DVD came into the market with unexpected velocity, many new surround formats appeared, and consumer buying habits also changed.

And so *The Rotel Encyclopedia* you're now holding in your hands contains some significant differences. The first is that we wanted to give consumers the same information that we formerly made available only to salespeople. In our view, knowledge truly is comforting – and the more knowledgeable both consumers and sales professionals are, the more satisfactory their collaboration as they assemble a home entertainment system or add a critical component to an existing system.

To that end, we've updated this book with a lot of new information – we've re-written whole sections and added others – to bring you the most up-to-date details we can.

We trust you'll find our efforts interesting and helpful.

Above all, we hope you enjoy this book. After all, that's what home entertainment is all about!

Thank you.

Written by:

TechniCom® Corporation
www.technicomcorp.com

Designed by:

Digital Arts & Sciences, Inc.
www.digiartsci.com

Illustrated by:

SLK Communications
E-mail: slkcomm@attbi.com

THE RIGHT STUFF . . .

Properly chosen and correctly installed, today's audio and home theater components put the excitement of the best in entertainment in our homes. In fact, if there's any one word that really describes – and sells – consumer electronics today, it is just that - "excitement."

Contrary to what you may think, technology is really secondary. To some salespeople – but certainly not the most successful ones – this may seem like heresy. But people know – instinctively, it seems – just how true this is.

. . . THE WRONG STUFF

We're not saying technology is unimportant. But today's components and systems offer benefits – how technology changes lives for the better, for example – rather than specific circuits that produce a particular effect.

WHAT YOU'LL FIND AND WHY YOU'LL FIND IT

We've re-written many sections of this Second Edition for both consumers and salespeople. But we've continued the very practical tone that made the First Edition so well received. Yes, you'll find a lot of technology here but we've intentionally focused on ways to make it more understandable and less intimidating.

We'll begin with stereo music reproduction. We'll look at what's needed to re-create all the subtlety, nuance, flamboyance, and just plain impact of live music in the home.

We'll examine certain design approaches that Rotel has found useful in designing equipment that accurately reproduces the emotional impact of great music or a great movie soundtrack. Judging by the number of awards and recommendations Rotel has merited in the last 40 years, we think you'll agree that these international reviewers and critics think we've been on the right track.

Then, because we're prejudiced in favor of good sound, we'll take an in-depth look at home theater with an emphasis on what's needed to get superb audio reproduction.

You'll find basic concepts explained in regular type. We've put more detailed information in **bold type** or in well-marked *sidebars*. Among other things, you'll find out:

1) **How to demonstrate (and how to listen to)** the excitement of good audio whether for music or movies.
2) **How components work together**.

3) **How to understand and explain the technology** in simple and non-confusing ways.
4) **How the audio side of home theater is similar** in many ways to good high-fidelity sound reproduction **and how it is different**.
5) **How to choose the proper system** that will satisfy a consumer's needs.

We've also included some **Demo Tips** we've found helpful in showing off good high fidelity and home theater sound as quickly as possible … preferably with a system a salesperson and a consumer have just agreed on. (Note: There's nothing tricky here. A good demonstration is structured to simply let you hear what's important.)

Each Section begins with a **Preview**. It's a short overview of the concepts you'll soon see in detail. We've also included a **Glossary** of commonly used terms. So, whether you're an audio professional or a consumer looking for some insights, a "warm fuzzy" or a "tweak freak," you'll benefit from reading this Guide.

One last note – obviously, we hope you remember Rotel as you make your selling and buying decisions. After all, we've spent many years designing and producing high fidelity and home theater components that redefine the word "value." So relax. Enjoy. After all, that's what this is all about anyway!

PREVIEW

Here's where we wave our flag. You'll find a lot of information about us, our attitudes, how we implement them, and other stuff you may or may not find important. We'll just let the facts speak for themselves.

Rotel is not a typical audio company. Unlike the corporate giants of the audio industry, Rotel is a family-owned business. We don't make video recorders, computers, or electric pianos. Instead, we've spent almost forty years building high fidelity and home theater components that meet two rigorous criteria: Musical accuracy and honest value.

Designed by an international team of experienced engineers, Rotel's prototypes undergo extensive evaluation and modification by some of the most critical ears in the business. Then our production engineering experts in Japan review the final circuits to make sure we can reliably manufacture these components in enough quantity to satisfy the growing number of Rotel customers.

After this critical phase, our quality-assured factories make sure that you get all the sound quality and rugged construction we've designed into every Rotel component. That's why we say that Rotel sets new standards in the pursuit of audio perfection.

OUR MISSION STATEMENT

At Rotel, we measure the quality of our audio systems not by mere specification, but by a higher standard: **Our goal is to reproduce an audio performance that can evoke the full spectrum of human emotion.**

Although this is an elegantly simple philosophy, it puts incredible demands upon our design and production people. But we think you'll agree that they've met the challenge.

EARLY ROTEL HISTORY

Rotel began its trek to world renown as a builder of quality audio components in 1961. During our early years, OEM (Original Equipment Manufacturing) assembly was our mainstay and accounted for nearly 70 percent of our business. Today, while we still have a share of loyal OEM customers, we focus our efforts on the Rotel brand. We want to become the best – not the largest – specialty audio company in the industry.

During the last several years, Rotel has achieved much of that goal. We've built a well-balanced product line and backed it with carefully considered distribution policies. However, our recent success is also derived from lessons learned in leaner years.

Rotel was shaped by many forces along the way, most notably industry overproduction of hi-fi in the late '70s. That global excess led to an ever-increasing trend towards feature-oriented product. Rather quickly, this evolved into a glitz marketing war. Instead of real performance evaluation through comparative listening, manufacturers sold products on such things as the lowest THD specification or the newest florescent bar-graph display. As you've already read in Mike Bartlett's welcoming letter at the beginning of this Guide, this weighed heavily on Rotel and caused us to completely rethink our product line.

By 1979, we had carefully reconsidered our strengths and decided to re-establish ourselves in the audio arena by going back to our roots. With help from a newly created British R&D team, Rotel began designing no-frills audio components that emphasized musical accuracy, superb build quality and affordability. This back-to-basics product plan was complemented by the factory's extensive experience in producing quality audio components reliably and inexpensively in the Far East.

This strategic shift produced a succession of highly praised models that were regarded by some of the most jaded British reviewers to be the yardstick of their class. In fact, the RA-820BX Integrated Amplifier won Rotel's first Product Of The Year Award in 1982 from ***What Hi-Fi*** magazine in the U.K. Every year since then, we've received a breathtaking number of awards from various magazines (well over 200 global awards to date) that cover every product category.

BALANCED DESIGN

Engineers can make many choices about how to allocate money while designing a product. "Price is no object" engineering is probably the easiest to implement. After all, if you throw enough money at something, you have a pretty good chance of building an exemplary product. Unfortunately, the price at which you must sell that product makes for a very narrow market indeed.

"Mass market" designs, on the other hand, provide a much larger window for consumer acceptance. However, the demands of this market call for extreme cost-conscious engineering to keep the units affordable. Unfortunately, to attract consumer interest these products tend to be feature-oriented rather then performance-oriented models.

Rotel doesn't follow either path.

Balanced Design is the essential Rotel philosophy. It's what makes us different and, at least from our point of view, what makes us better.

Balanced Design is a disciplined synthesis of physics, electronics, and mechanical engineering guided by the following beliefs:

1) Great sound need not be the most costly.
2) No single design aspect will be emphasized at the cost of lower performance elsewhere in the product.

Balanced Design revolves around three major areas: **Parts selection**, **circuit topology**, and **critical evaluation**. These aspects influence final sound quality significantly and illustrate Balanced Design's holistic approach to product development.

Better Parts

Consider this example of Balanced Design in action: A standard low-cost electrolytic capacitor may well help filter unwanted current ripples from the power supply. However, we know there would be significant negative effects on sound quality if we used one of these "good enough" off-the-shelf circuit elements. Instead, Rotel engineers select parts to give our components the best reliability and acoustic advantage. They listen to new model prototypes and revise production plans when they feel they can gain a sonic advantage by using a particular component from another manufacturer, or by specifying closer tolerances. We source parts from highly respected suppliers around the world, including many of the following specialty manufacturers: BHC "Slit Foil" and T-Network electrolytic capacitors from England, Rubycon "Black Gate" electrolytics from Japan, LCR polystyrene capacitors from Wales, Vishay (Japan) and Roederstein (Europe) 0.1% precision metal film resistors, Alps pots from Japan, and high-performance Signetics, Analog Devices, and Burr-Brown ICs from the US. These are the same high-performance parts that many esoteric brands underscore as part of their claim to better sound.

Circuit Topology

It's not enough to simply use good parts. Where you put them is equally important. That's why we keep circuit paths as short as possible – there's less chance that spurious emissions and unwanted leakages will corrupt delicate signal nuances. And that's why we often use more expensive "on-board" relay switching rather than routing a signal from a rear panel input to a front panel control, for example. That's also why we rely heavily on something we call Symmetrical Signal Trace design. This keeps each channel's signal path identical to the others to preserve imaging and soundstaging. We also use superior "star" grounding techniques to improve performance because these route all ground connections to a single point and reduce potential loops that might produce hum. Further, our amplifiers don't use output inductors because they decrease control and, consequently, sound quality.

Critical Evaluation

For most companies, the right parts in the right places would be enough. Not for Rotel! Over the years, we've found that human hearing is often more sensitive than even sophisticated instruments in the hands of experienced engineers. That's why our development team places such importance on listening. Actually, "listening" doesn't quite describe the process, which involves specially trained people in our organization with extraordinary aural acuity to attend pre-production audio evaluations and approve the design or send it back to the drawing board.

These evaluation sessions take place many times during our product development efforts. Sometimes we listen to different circuit elements – capacitors, resistors, and active devices like transistors or operational amplifiers – before making a choice. At other times, we listen for differences in circuit layouts or differences caused by power supply elements. We note these differences carefully as we choose the most effective parts for each of our components. This whole process is far more than just "lip service," too.

On occasion, we'll change pre-production samples when it's obvious that modifications will improve sound quality. We sometimes even change production schedules so we can get better parts for critical circuits. Although this may cause our global markets some pain, we continue to do it. After all, a better sounding component a month late is, at least in our opinion, much better than a less revealing one on schedule!

Summing Up

Bob Tachikawa, Rotel's senior executive and a graduate engineer himself, put it this way:

"We pick individual components only after we've compared what's available. We select every part according to a rigorously applied high-performance/fair price formula. That's why we don't use something just because a 'sister company' developed it and why we don't need to develop leading-edge technologies – we just make them **sound** ***better."***

How do you benefit from Balanced Design? We think ***Audio Adventure*** magazine (USA) said it best when presenting us with their very first "Manufacturer of the Year" award in 1996: "Rotel's products, at all prices, are so impressive that it is safe to think of Rotel as a 'gold standard' against which we can confidently measure the industry's performance as a whole."

OUR ENGINEERS

Balanced Design didn't come about by accident. It evolved because our engineers are, first and foremost, music lovers who labor over their designs like proud parents. They listen to the results, then tweak and adjust circuitry until the new product meets the team's exacting musical standards. Of course, when you remember that Bob Tachikawa, the head of the company and son of Rotel's founder, is himself a graduate engineer and an audiophile, this attitude shouldn't be unexpected, should it?

PERFORMANCE AND VALUE

Make no mistake, the "price/performance ratio" is a constant engineering challenge. Building a range of electronic components that provides reliable and affordable high-end performance has kept many an engineer up until the wee hours of the morning.

But this is Rotel's greatest strength. Simply stated, Rotel provides the best value in conventional stereo and home theater components. It's the specialty dealer's step-up line that bridges the performance gap between mediocre hi-fi/home theater and "I don't care what it costs, just give me the best!" electronics.

CRITICAL PRAISE

We've earned our reputation for excellence. Just look at the hundreds of rave reviews and awards from respected reviewers who listen to music every day. Hardware is secondary to these seasoned veterans; faithful reproduction of the original musical performance is their standard of measure. Comments bestowed by these keen ears have kept us true to our goal – the pursuit of hi-fi equipment that is musical, reliable, and affordable.

POINTS OF DIFFERENCE

Every manufacturer talks about "product differentiation." And there's nothing mysterious about this term either – it simply means all the things (features, parts, design philosophy, etc.) that make a product different from a competitor's. With Rotel, that difference is simple. Here are some of the points we feel are important:

1) Rotel is primarily a step-up "separates" line.

Even though we design, manufacture, and sell outstanding A/V receivers, for example, we still strongly recommend that you consider the benefits of separate components.

- An A/V receiver, although it answers many needs in a relatively compact package, does not provide an easy "upgrade path." Reconfiguring the system or adding power is often more difficult than it would be with separates. If a receiver offers convenience, separate components offer flexibility.

- Another obvious benefit is improved performance. It's the direct result of separate, isolated power supplies for each unit. When a tuner, preamp, and power amplifier all share the same power supply – as they invariably do in a receiver – there are complex demands constantly placed on that supply. This ever-changing load is like having multiple water faucets open in your home and then wondering why you can't get a good, strong shower at the same time!

2) Rotel is a full-line supplier of quality, specialist audio and home theater components.

- We're not just a "Hi-Fi" manufacturer. Our product line includes high-quality home theater components like DVD and DVD-A players, multi-channel power amplifiers, remote-controlled surround sound preamp/processors with DTS and Dolby Digital capability.
- In addition, we now offer multi-zone systems designed specifically for the custom installation market. Why? That's simple: There's a real need for a custom system that sounds great, is easy to sell, install, and use.
- Other manufacturers or marketing organizations that attempt to answer the needs of the performance-oriented customer most often fall into the "bits and pieces" category with a preamp here, a processor there, but no consistent, systematic approach to product development. At the other extreme, they may try to be all things to all people, and end up with such complicated products that a minor problem often results in total system reprogramming. Neither of these descriptions fits Rotel.

3) Rotel makes it better . . .

- Here is where Rotel stands apart from many "manufacturers" who only draft rough design specs and then farm out the actual production engineering and assembly to a remote factory. This puts them at the mercy of the manufacturer's production schedule and quality control – often with less than stellar results.
- In contrast, Rotel manufactures and directly maintains quality control of all of our products "in-house." That means we invest our pride in assembling every product that bears the Rotel name. That's why we enjoy substantial advantages in achieving musical accuracy and maintaining long-term reliability.
- Rotel meets or exceeds demanding world-class standards covering everything from management techniques, incoming parts inspection, test instrument calibration, assembly tolerances, and final quality control inspections.
- In addition, we test components extensively at our U.S. National Service Center in North Reading, Mass. This gives us immediate "in the field" experience with everything we manufacture and assures you of quick diagnosis and fast turn-around in the event something does go wrong.

4) . . . so Rotel backs it better.

- We know our products are reliable. That's why we have one of the longest warranties in the business – five (5) years* parts and labor on all electronics and two (2) years* parts and labor on mechanical devices like DVD and CD players, and cassette decks.
- In fact, Rotel is nationally recognized for its superior service. ***Inside Track*** (USA), a well-known industry newsletter, polled many dealers for comments. Here's a sample: "This is one company whose product we just don't see back."
- No, we're not claiming to be perfect – we have occasional problems. But we handle them quickly. To quote another dealer: "The thing I really like about that is all you have to do is call up . . . You get an RA instantly and the unit is here in a couple of days. The support and service are excellent." This sentiment has remained constant as evidenced by the fact we have stayed in the top tier of ***Inside Track's*** annual dealer survey ever since.

* U.S. warranty, Other countries may vary.

PREVIEW

Even if you're not a "techie," you'll enjoy this section. You'll find a number of important circuit concepts explained clearly and completely.

At the end of this Section, you'll know about power supplies, high points of analog circuits, and the answers to lots of those confusing questions about different approaches to digital circuit design.

Whether you're a salesperson or a consumer, we think you'll appreciate the clear yet complete discussions in this Section. The result? If you're a salesperson, you'll benefit from the clear descriptions. You can use many of the points here to refine your presentations so they'll be more authoritative and confidence-inspiring. And, if you're a consumer, you'll have the information you need to ask the kinds of questions that'll help you focus on the best equipment for your needs.

POWER SUPPLY HIGHLIGHTS

Introduction

Make no mistake about it, the power supply is the backbone of any component. Power supplies do several things:

1) They **convert wall-socket AC** (alternating current) into lower voltages that can be used by individual circuits. Most power supplies use a transformer, specifically a ***step-down*** transformer, to convert either 120 volt or 220 volt current to more useable 30 or 50 volt levels.
2) They **transform AC voltages into DC (direct current)** by passing the output of the transformer windings through devices called ***rectifiers***.
3) They **stabilize DC currents** by passing them through ***voltage regulators***.
4) They **store DC energy** in ***capacitors*** until it's needed by different circuits. Although this seems comparatively simple, the problem is that good power supplies are also expensive power supplies. This is especially true for power amplifiers where the transformer is usually the single most costly item in the chassis! Another complicating factor is that, while everyone talks about power supplies, relatively few actually understand what they do (present company excluded, of course!)

Some manufacturers use this combination of expense and ignorance as an excuse to specify a barely adequate supply. After all, they ask, who'll really know?

You will. Oh, you might not instantly identify the power supply as a reason why one component sounds rich and robust while another sounds anemic and strident but you'll certainly hear the difference. No, a power supply is not the ***only*** sonically critical factor in a component's design, but it is very important.

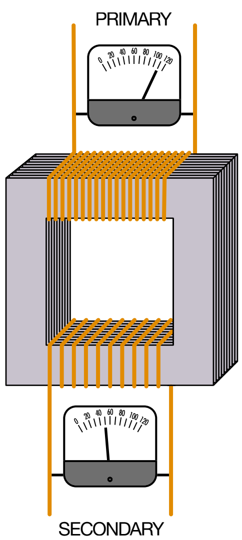
Let's take a closer look at some of the elements we identified a few paragraphs ago.

Transformers

In its simplest form, a power supply transformer consists of two closely spaced coils of insulated wire wrapped around a magnetic core or former. One coil (the ***primary***) connects directly to a voltage source (120 volts from a wall socket in North America, 220 volts in Europe and other parts of the world). As it flows through the primary, this current induces a voltage in the second coil (called, appropriately enough, the ***secondary***.) Although the induced voltage in the secondary coil can be the same as the primary's direct voltage, it most often is lower – thus the term "step-down" transformer. Designers choose secondary coil voltages by varying the number of windings (or turns) of both primary and secondary coils.

A secondary's full power potential (voltage and current combined) is measured in voltamperes (abbreviated VA).

A transformer can have more than one secondary coil. In fact, this is rather common in audio and home theater components. Two or more secondaries can provide different voltages for different circuitry blocks. In addition, the fact that there's no direct contact between primary and secondary coils means that a transformer provides a certain degree of isolation. (That's one of the advantages of "induced" voltages.)



Simple transformer with 120 volt primary and one 50 volt secondary.

Isolation has two main benefits:

1) Transformers reduce RF (**R**adio **F**requency) and EM (**E**lectro**m**agnetic) interference that usually contaminate the AC drawn from a wall socket. (Some transformers also include electrostatic shields that further reduce interference.)
2) Multiple secondaries reduce unwanted interaction between various circuitry stages – the digital and analog sections of a DVD or CD player, for example.

In addition to circuit considerations (number of secondaries and their induced voltages, etc.), a transformer's very shape also has an impact on effectiveness and efficiency.

Let's look at two popular types; the "EI" and the toroid (or toroidal) transformer. Both types are reasonably efficient. They provide close spacing between primary and secondary coils and both place the coils relatively close to the magnetic core.

Toroids have one major advantage. Because of their donut-like shape, they tend to "focus" electromagnetic fields more tightly and produce slightly less spurious radiation than their EI cousins. This can mean that a power amplifier, for example, might be slightly quieter if a toroid is selected.

In tight spaces, however, the EI design is actually better. It's more compact and requires less mounting space than a toroid of equivalent capacity. The only caution is that the designer must place the EI transformer properly as this has an influence on its magnetic flux line orientation: Angled improperly, an EI core design can actually increase circuit noise.

Rectifiers

Rectifiers convert AC (alternating current) into DC (direct current) pulses. In short, they're the electronic equivalent of a one-way street. Put an AC voltage on one side of a rectifier and you'll get a series of DC pulses at the other. (Don't lose any sleep trying to figure out how this happens.)

Whenever engineers talk about rectifiers, they use terms like "forward voltage drop," "reverse resistance," "maximum peak current," etc. These terms simply mean that rectifiers must be able to handle the voltage and current supplied by a transformer's secondary coils. Fortunately, picking the proper rectifier isn't difficult – you just need to make sure you've got sufficient capacity to handle the coil's output.

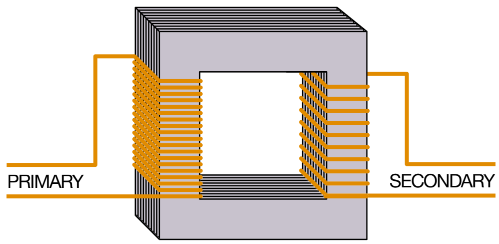
Although you may see terms like "full wave rectification" and "half wave rectification" applied to the appropriate stages of a power supply, don't worry. This bit of esoterica isn't a major consideration for our purposes.

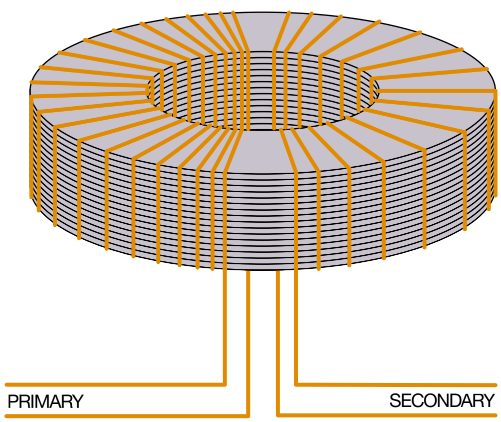
Regulators

As their name implies, voltage regulators simply keep DC operating voltages within a narrow specified range. They function as comparator circuits and generally use zener diodes as reference sources for a "pass device" that assures proper circuit functioning under a variety of operating conditions.

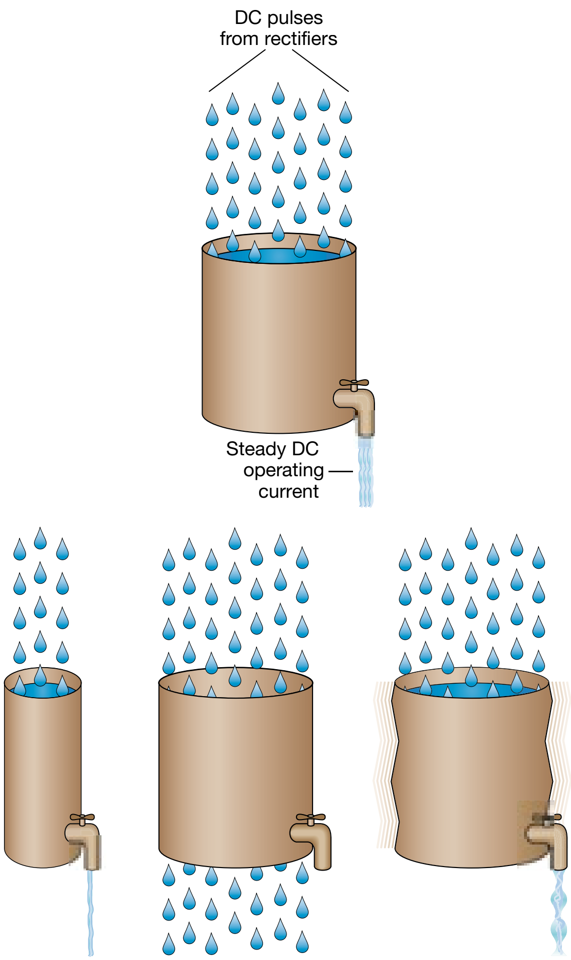
Regulators take many forms. Some are defined by function, series- and shunt-types for example. (Again, don't worry about these terms – we're just mentioning them here so you won't be surprised if you see them in some brochure.) If you see the words monolithic or hybrid appended to "regulator," don't worry. They simply describe how a regulator is made. Again, it's something you needn't lose any sleep over.

You'll probably see comments about "fully regulated" and "unregulated" power supply designs, particularly in reference to power amplifiers. Again, keep sawing those "Zs." The arguments for and against each are long, involved, and usually unresolved. Listening to music is much more enjoyable than paying attention as two engineers have at it!





This schematic representation shows the basic physical differences between an EI and a toroidal transformer. The toroidal produces a more focused "hum field" that helps reduce noise in surrounding circuitry. However, the EI type is smaller, less expensive, and provides good performance when properly engineered into the design.



Just as the physical properties of the bucket catching the rain drops affects the flow of water out of the spigot, the properties of the capacitors used in electronic circuits affects the quantity and quality of the current flow from them.

Capacitors

Capacitors are energy storage devices composed of a rolled conductive plate sandwiched with an insulating material called the dielectric. Think of two sheets of aluminum foil separated by a sheet of plastic refrigerator wrap. If you roll these three sheets into a tight cylinder, you'll have a capacitor. (Well, sort of . . .)

As used in power supplies, capacitors convert the DC pulses produced by the rectifiers into steady-state DC current. To understand this better, imagine a bucket with a spigot at the bottom sitting in a rainstorm. It collects raindrops (the DC pulses) but releases a steady stream of water (steady DC current) from the spigot. It's that simple – at least in concept!

In actuality, however, things are more complex. Capacitors have many characteristics that influence circuits. This is due, in part, to the fact that there are many different types of capacitors.

Consider, for example, a bucket with a wide top and very small spigot, or one with a very narrow top and a very generous spigot. Imagine a bucket with no bottom at all or one made of a highly elastic substance that trembles every time a drop or water hits the exposed surface of the pool and how these vibrations actually change the intensity of the flow of water from the spigot! In electronic terms, all capacitors have a certain ability to store energy. This is usually stated in microfarads (abbreviated µf) referred to a specific voltage. You'll find these numbers stated on the capacitor body.

In addition, all capacitors exhibit other characteristics as well: inductance, resistance, and dielectric absorption. Three other attributes, dissipation factor (DF), quality factor (Q), and equivalent series resistance (ESR), also affect performance. DF, Q, and ESR all represent heat-producing losses within the capacitor. Minimizing these losses and the heat they produce results in greater stability, better reliability, and reduced distortion.

Capacitors are made of different materials and in different ways, depending on their application and this accounts for the significant difference between power supply and signal path caps. However, different capacitor types used in the same place affect sound quality too. For instance, Rotel uses specially designed Slit Foil capacitors which enhance power supply "speed" and purity by acting as multiple, smaller value capacitors with much lower ESR for effortless music reproduction.

In addition to the Slit Foil capacitor, Rotel is the first company to use an even newer development, the T-Network Capacitor, or TNC, in many of its products. Comparing the TNC with several different electrolytics reveals sometimes substantial differences. Less expensive caps generally produce grainier, rougher sound with particularly "spitty" highs. In addition, these "run of the mill" caps evidence softer focus, a somewhat forward midrange and less well-defined bass.

Moving up scale by substituting TNCs improves things a great deal. Among other things, the TNC seems to enhance timbral accuracy, image depth, and transparency. Other benefits include a clearer focus as well as a less etched quality to high-frequency reproduction.

The T-Network Capacitor doubles the connections to other circuit segments for better "speed" and resolution.

ANALOG CIRCUIT HIGHLIGHTS

We'll approach this rather daunting topic from two viewpoints, the theoretical and the practical. In the theoretical section, we'll highlight some guidelines we've found invaluable in designing award-winning components. (But don't worry – we won't get ***too*** technical!) The practical section will show you how Rotel products benefit from our beliefs.

Theory

Rule # 1: KISS (Keep It Simple, Stupid!)

Mark Twain, one of America's best writers, once said to a friend, "I would have written a shorter letter but I didn't have the time." He was referring, of course, to the challenge of simplicity, of describing things accurately yet succinctly. That's a challenge Rotel's engineers take seriously indeed. In fact, if any one word describes Rotel's approach to circuit design, that word is "minimalist."

This simply means we always opt for fewer parts and shorter signal paths, even though these circuits usually take longer to design! This approach greatly lessens chances for signal distortion and interference.

Rule # 2: The Good Stuff, Please . . .

Of course, good analog circuits, even our minimalist designs, need the right parts. All the individual components that make up the final circuit must have the precisely calculated values specified by the design engineers. But "paper specs" are just the beginning.

Rotel engineers and listening experts evaluate many elements – different brands of bipolar output transistors with identical performance parameters, for example – to find the best one for a particular application. This meticulous yet cost-conscious auditioning process adds real value to Rotel designs.

Rule # 3: . . . But Only In The Right Places!

Next comes the even trickier part of selecting just which element melds best with others to produce the best ***sounding*** circuit. For example, a metal film resistor in place of a conventional carbon resistor may have a major impact on sound quality at one critical circuit junction but almost no effect at another! A "cost-no-object" design would simply use metal film resistors everywhere but that brute force method doesn't result in affordable components. The trick is knowing where to use the good stuff and where it's simply a matter of overkill.

Rule # 4: Simple On Paper, Simple On the Circuit Board

Then comes yet another challenge: Arranging the most compatible parts on a circuit board so that they function optimally together; routing ground paths so that they don't add noise to the audio signal, shielding low-level signals from transformer hum fields, etc.

This careful signal path mapping is called topology and it's a critical design phase for Rotel. Although we make extensive use of computerized circuit modeling and routing techniques, we know there's no real substitute for the intuitive understanding of these subtle relationships. Rotel customers reap the sonic rewards that come from our engineers' experience in designing literally hundreds of successful components.

An example is Rotel's Symmetrical Signal Trace strategy, which makes each channel's signal path either identical to or a mirror image of its twin. This ensures absolutely convincing stereo imaging by preserving the precise intra-channel time relationships so essential to proper spatial reproduction.

"Star grounding" is another example. Here, all circuit grounds go to a common, centrally located position. This drastically reduces the chances for "loops" that produce excess hum and noise.

Practice

To see how well these ideas survive in the real world, let's look at a multi-channel power amplifier.

Notice the power supply transformer in the center of the chassis. This custom toroid design delivers 1.2 kVA (1,200 volt-amps or watts) from multiple secondary windings for enormous reserve current capability. As if that wasn't enough, note that there are *two* such transformers, one mounted above the other, that serve as the heart of this amp's power supply.

This true high-current supply won't run out of juice even under the most demanding conditions. Although this might be difficult to see in the photo, you won't find any "chokes" on the output sides of the transformers, either. These chokes, often used in less capable designs, artificially restrict current to compensate for inadequate output stages.

NOTE: If you see a "speaker impedance" selector switch on the back panel of a competitive component, beware! This tells you that the transformer or output stage – or both – is a "low-current special." These switches usually insert resistors to cut current flow from the transformer's secondary windings. They're needed because some (usually inexpensive) transformers incorporate marginal secondary windings that cannot supply more than nominal amounts of current without overheating.

Using advanced computer-aided design tools, Rotel engineers pay particular attention to circuit board layout.

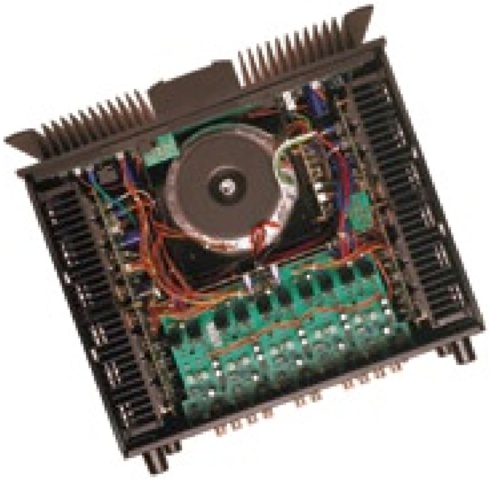
The eight storage capacitors used in this amp (they're "buried" under the circuit board at the rear of the amplifier) are special high-capacity BHC Slit Foil designs that act as multiple, smaller value capacitors for exceptional purity and effortless music reproduction.

The output stages themselves are a textbook example of proper high-current execution. Although the amplifier is rated at "only" 200 watts per each of its five channels, it has thirty 150 watt, 15 ampere output devices! If you add these figures, you end up with a maximum theoretical output capability of over 4,500 watts at 450 amperes! How's that for overbuild?

Of course, practical considerations make this theoretical output rather difficult to achieve but you get the idea! Actually, the real-world benefit to this configuration is the enormous S.A.O. (Safe Area of Operation) that adds to the amplifier's stability and longevity, two attributes often overlooked in less-careful designs.

High Current - The Real Story

Let's take a closer look at something many manufacturers talk about but few really deliver: High current design. Of course, the concept "high current" can apply to almost all analog circuits but we'll use the phrase to describe one approach (and the correct one, we firmly believe!) to power amplifiers.



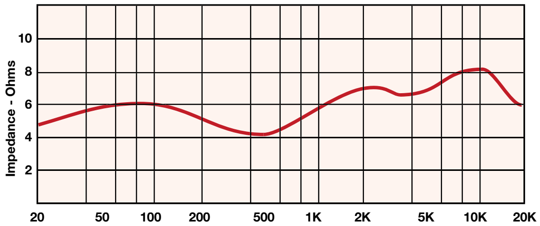
Here, a Rotel power amplifier illustrates the clean, uncluttered layout and premium components that contribute to superb sonic performance.

Let's begin by asking why high current capability is important. The answer to that is simple: Many of today's speakers are not "amplifier-friendly." Although they may sound great, they really tax an amplifier's ability to get the most out of them.

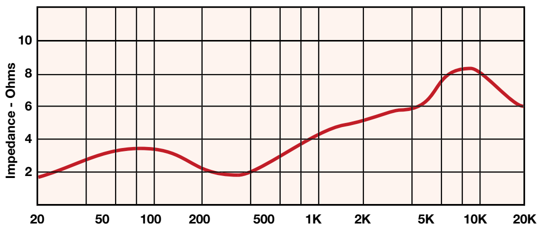
High current amplifiers deliver the sound, low current designs don't. It's that easy. The reason is fairly simple. Engineers design amplifiers (or any circuit, for that matter) with a particular "load" (or resistance) in mind. With amplifiers, that "load" is the loudspeaker.

A NOTE ON IMPEDANCE

A loudspeaker's impedance (resistance or "load") is measured in ohms. Higher resistances (6 to 8 ohms) are usually easier for conventional solid-



Loudspeaker A



Loudspeaker B

Loudspeaker B is actually more difficult to drive than loudspeaker A because the low impedance in the bass frequency range demands far more current from the amplifier.

state amplifiers to control, so you might think a 4 ohm speaker is always a more difficult load than an 8 ohm design.

Sorry, but this isn't necessarily true.

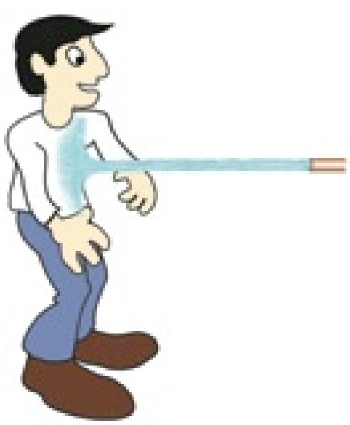
You see, impedance varies with frequency and a speaker rated at 4 ohms might just be a more benign design than another with a "nominal" 8 ohm rating that averages out large variations. Here's a comparison.

So, before you make a quick judgment based on "nominal" impedance ratings, know that this isn't the whole story. You really need more detailed information! Of course, if your amplifier delivers lots of current at low impedances in the first place, this is really a moot point!

Note: Here's where things get a bit "techie." But bear with us — we'll give you a "real world" explanation shortly.

We're all used to amplifier measurements of "100 watts," " 250 watts," etc. However, we often forget that this measurement — even when it conforms to the FTC (Federal Trade Commission) format that requires accompanying impedance, bandwidth, and distortion data — doesn't tell us everything we need to know.

If you want a crude analogy, think of two water pipes, one 2" in diameter, the other 12" wide. Even when the water pressure in each pipe is identical, the larger pipe would, as you can easily understand, carry more water in a given instant in time. It's simply bigger.



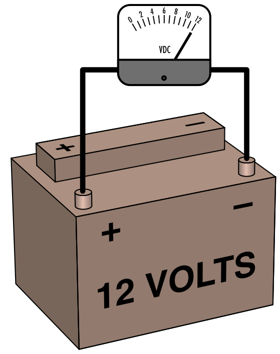
Even when the water pressure (voltage) is the same, the larger pipe transfers more power because of its larger diameter (current capability).

Now, let's apply this reasoning to amplifier design. If we think of the ***water pressure*** as ***voltage***, and the ***volume*** of water through each pipe in any given instant as ***current***, the implications become obvious. Just as the 12" pipe can move more water than the 2" pipe can, a high current amplifier can deliver more power than a low current design.

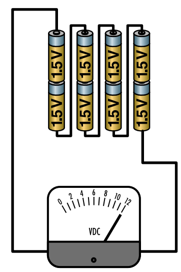
Now, let's examine this from the vantage point of Ohm's Law which says, in part, that power is proportional to the *square* of the current. In other words, a small increase in current capability may pay large dividends in an amplifier's real-world power transfer capability.

Realize that wattage measurements alone do not tell us how much *current* an amplifier can produce. That's why two amplifiers, each rated at 100 watts per channel, might be very different in their ability to deliver real power to low impedance loudspeakers. That difference is *current*.

Here's another example. Compare 8 AA flashlight batteries to a typical car battery. Combining all eight AA batteries will give you 12 volts (1.5 volts for each battery x 8 = 12). The car battery is also rated at 12 volts, right? So here's the question – could your 8 flashlight batteries start your car's engine on a cold day? Of course not! (Although you may want to try this yourself, we think you'll probably just wind up with a lot of dead AA batteries — and be *very* late for work in the bargain!)



That's because the engine is a far heavier "load" than a flashlight bulb. So the motor demands far more current. And that's the major reason a car battery is far larger than 8 AA batteries!



In a similar manner, a low impedance loudspeaker is a more difficult "load" than a higher impedance speaker. Low impedance designs demand more current from the amplifier and sound much worse if the amplifier can't deliver it. This problem really intrudes on accurate reproduction when you turn up the volume or play music or a movie soundtrack with a lot of bass energy.

So, where does real current capability come from? Simply put, high current capability results when an amplifier combines a substantial power supply with enough output devices to deliver relatively unrestricted current flow to a loudspeaker regardless of its impedance.

THE POWER STORY: WATT'S IT ALL ABOUT?

"I want a 100 watt amplifier!"

How many of us, salespeople and consumers alike, have voiced that sentiment? Probably quite a few. In fact, that measurement has become a *de facto* standard today. But how many of us really know how to interpret this specification in real world terms? Probably not many! So here's some thoughts that should help.

At The Beginning

The "watt" is the single most important measure of an amplifier's ability to provide sonic enjoyment. But it's also the most widely misunderstood term in home theater and high fidelity.

A *watt*, named after the British scientist James Watt, is a *unit of power*. Since power is always expressed as "a unit of energy divided by a unit of time," we define a watt as "*a unit of power equal to one joule per second.*"

In more practical terms, a watt is a measure of an amplifier's ability to perform work – to move the speaker diaphragms that produce sound.

Viewed this way, watts are similar to horsepower. In fact, there's a direct relationship between them – one horsepower is equal to 746 watts. (For the curious, one horsepower is the power needed to lift 33,000 pounds a distance of 1 foot in 1 minute.)

So, if you ever hear of a "two horsepower amplifier," you now know that it would pump out 1,492 watts, thus making it an ideally attractive item for a Columbus Day sale event!

"Go Fast" And "Get Loud"

Because most of us think of watts as "electronic horsepower," we also feel that the more of them, the better. After all, if more horsepower in an automobile means quicker acceleration, then more watts in an amplifier means louder sound, right?

Unfortunately, that's not always true.

Just as a 200 horsepower engine will provide varying degrees of acceleration depending on the weight of the car it has to pull – or push – around, a 100 watt amplifier will provide varying degrees of loudness depending on the *efficiency* of the speaker connected to it. Other factors, particularly the size and acoustic qualities of the room housing your speaker, also influence loudness. But efficiency heads the list.

This means that wattage ratings alone may not be a clear indication of an amplifier's ability to satisfy your needs.

Power Factor

The exact relationship between amplifier power and loudness is difficult to understand. Although more watts *can* mean more loudness, there are some variables involved.

To better understand this, let's start with the *decibel*.

A decibel (abbreviated dB) is the smallest difference in loudness most people can perceive. In other words, if you slo-o-o-o-w-ly rotate a volume control (either up or down) until you can just about make out a level difference, you've probably increased or decreased the overall playback level by just 1 dB.

A note: If we were being strictly accurate, we'd use the term "dB SPL" here with "SPL" standing for Sound Pressure Level. There are many other qualifiers an engineer tacks on to "dB" depending on how they're using the term. Trust us when we say that you don't need to know anything about them right now!

If you want to increase the volume so that the loudness change is obvious, you'll find that you've increased that average level by 3 dB. And if you want to make the playback level *twice* as loud (subjectively speaking, that is), you'd need to increase it by a measured 10 dB.

This brings up the real question – how much more amplifier power will those increases require?

Before you try to answer, here are some basic rules of thumb:

1) If you want to raise the average playback level by just 3 dB (a "readily apparent" difference), you need to *double* the amplifier output
2) If you want to *double* the playback level (in other words, to raise it by 10 dB), you'll need an amplifier with 10 times the output capability of your previous unit.

In other words, if the reason you're thinking of changing amps is to put more "umppff" in a system, you'll need to begin with an amp that's at least *twice* as powerful as the one you have.

And if you really want to rattle your neighbor's walls, you'll need a powerhouse with at least 10 times the output capability of the present unit.

Think about this for a moment. If your current amplifier is rated at 50 watts per channel and you want a bit more volume, you should start looking in the 100 watt/channel range. Anything less powerful simply won't give you the improvement you want. And if you really want to "run with the big dogs," you're going to need a 500 watt/channel super-amp!

Sobering, isn't it?

The Real World Gets Worse

Of course, a super-amp alone does not a louder system make. Why? For one thing, we need to know the power-handling capabilities of the speakers connected to it. After all, it would do little good to sell or buy a "super amp" just to produce more volume only to find out that the speakers couldn't handle the power!

In simple terms, power-handling capacity simply means the amount of amplifier power a speaker can absorb before it stops working. The problem is that speakers are notoriously inefficient. On the average, they convert less than 3% of an amp's power into sound. The rest ends up as heat that eventually roasts the speaker's voice coils.

When we try to measure power-handling capacity, we run into a severe problem: There is no single standard that's been accepted by the entire industry! (That's right – all those "recommended power range" specifications you see on a speaker spec sheet are, at best, educated guesses!)

A good power-handling spec would tell you three things:

1) The type of test signal being used (narrow-band, broadband, sine wave, square wave, etc.)

2) The power level measured in RMS watts
3) The length of time the signal was applied to the speaker

In addition to power handling-capacity, there are two other factors we need to consider. They are:

1) The loudspeaker's efficiency
2) The difference between average and peak sound pressure levels.

Most people are a bit confused by the concept of a loudspeaker's efficiency. The term is simply a shortened form of "conversion efficiency" – or the percentage of an amp's electrical power that actually gets transformed into acoustical power we can hear.

As we've already said, loudspeakers are horrendously inefficient. They convert, at most, 3% of an amp's power into sound with the remaining 97% ending up as heat! As you can imagine, manufacturers are hesitant to publicize a product this way so they've adopted a "sensitivity" rating instead.

This sensitivity spec gives you an idea of how much sound you can expect from your speaker.

THE SENSITIVITY SPEC EXPLAINED

When you see a sensitivity measurement like "86 dB SPL with 1 watt @ 1 meter" on a spec sheet, here's what it means: "When we apply a random noise signal measuring 2.83 volts at the speaker terminals and measure the loudspeaker's output with a calibrated microphone placed 1 meter away from it, we'll get a sound pressure reading of 86 dB SPL."

(Note: 2.83 volts is simply the equivalent of 1 watt across a resistance of 8 ohms. Don't worry about the fact that some speakers have a different impedance rating (6 ohms, 4 ohms, etc.) The "2.83 volts" is simply an engineering convention that takes those differences into consideration.

Here's a somewhat arbitrary hierarchy of relative sensitivity ratings:

84 dB or lower	*Low efficiency*
85 – 88 dB	*Moderate efficiency*
89 dB and higher	*High efficiency*

Some people will argue that higher sensitivity ratings are intrinsically superior. That's nonsense. However, a higher sensitivity rating does mean that a speaker will generate higher sound pressure levels with the same amp that might only produce moderate levels with a low-efficiency design. The other side of the coin, obviously, is that a speaker with a higher sensitivity rating may well provide all the level you want with a very modest amplifier indeed.

Let's apply our "Power Factor" to the sensitivity spec:

Desired sound pressure level	Required Power
86 dB (moderately loud music)	1 watt (that's just what the spec tells you.)
89 dB (a fairly audible increase)	2 watts (double the power)
96 dB (loud music)	10 watts (ten times the original power)
106 dB (really loud music)	100 watts (ten time again)
116 dB (the onset of hearing damage)	1,000 watts (another ten-fold increase)

As you can see, when we begin with moderately loud music, the power demands go up substantially as we give a clockwise twist to the volume control.

Keep in mind that these requirements are for average levels only. When we talk about peak levels like cymbal clashes or explosions, the power requirements are much greater – sometimes more than 100 times the average output power! And don't forget that one of the most important aural differences between live music and "canned" reproduction is the dynamic range of the sound we're listening to.

This is where larger amplifiers come into their own. It's not *average* playback levels that distinguish capable amplifiers from the also-rans but *peak* level capability. That takes *real* watts – and lots of 'em!

Watts: Meaningful Measurements

This is the single most important question you need to be aware of. Amplifier power specs can be misleading, particularly in today's world of multi-channel sound. Here's some information to help you evaluate your choices.

In the early days of high fidelity, wattage ratings were very confusing. Forget about the differences between RMS (average) and Peak Power ratings – they were minor. Audio salespeople used to joke about fictitious power "standards" like JBIE (Just Before It Explodes) and ILS (If Lightning Strikes).

Things got so bad that the Federal Trade Commission ruled that all claims had to include the conditions under which power was measured. Specifically, the FTC said:

1) All power measurements had to be in RMS (or average) watts.
2) The band of frequencies ("bandwidth") for which the measurement was valid had to be stated.
3) Total harmonic distortion (THD) figures had to be supplied.
4) The load (or impedance) had to be specified.
5) The number of channels being "driven" into that load had to be specified.

Based on those standards, here's a valid power specification: "100 watts RMS per channel, all channels driven into 8 ohms from 20 Hz to 20 kHz with no more than 0.08% THD."

Today, when multi-channel amplifiers are pushing way past the performance envelope possible when the FTC stepped in, these parameters still have value. In fact, we feel very comfortable in saying

1) A*ny power spec that does not include all these qualifications is somewhat misleading*
2) *Even when a spec is complete, you'll still have to read it carefully to make sure you're getting what you* think *you're getting!*

Notice that the FTC did *not* say that everyone had to compete on the same "level playing field," only that everyone clearly state what kind of a "home field advantage" their power measurements might give them. That's why we're even more cautious with something like "100 watts per channel *per FTC regulations*." That tells you little other than the fact that someone is probably trying to pull the wool over your eyes!

Let's take a closer look at the five conditions you'll find in any good power specification.

1) "RMS" stands for "Root Mean Square" and it signifies a precise, mathematically correct way of expressing useable *average* power output rather than some instantaneous "peak" measurement that bears little relation to real world operation.

2) "Bandwidth" tells you the range of frequencies over which the amplifier can produce the claimed power.

   An amplifier has more trouble producing power at the frequency extremes than it does in the midrange. Bass reproduction in particular requires a beefier, more stable power supply. Although all of us vary a bit, most of us can (or could, depending on our age) hear frequencies from as low as 20 Hz to as high as 20 kHz. Because you don't know the exact frequency range of your favorite music or soundtrack in advance, it's comforting to know that your amplifier will be able to reproduce buffalo bellows and bat beeps with equal aplomb.

   Any time you see a power rating that says "*x* watts @ 1 kHz," look out. That rating tells you very little about the amp's ability to deliver the low-frequency goods. Yes, these "1 kHz" ratings may be legitimate in that they adhere to the DIN standard popular in Europe, but the fact remains that the DIN ("Deutsche Industrie Normen" or German Industrial Standard) is far less demanding than our FTC-originated spec.

3) Distortion: This is more of a mixed bag than it appears to be at first. When the FTC stepped in, Total Harmonic Distortion (THD) and Intermodulation Distortion (IMD) were the generally agreed-upon standards. But both are so-called "steady-state" measurements whose relationship to real-world listening is not always clear. For instance, neither measures an amplifier's behavior when reproducing the intense peaks we often encounter in music. And both forms of distortion can be lowered with large amounts of "negative feedback" in a circuit. Unfortunately, this technique often has a less-than-wonderful effect on sound quality even as it reduces steady state distortion.

   Distortion measurements have value, however. They tell us when an amplifier is operating in a non-linear fashion; how it is changing (distorting) an input signal. Higher distortion measurements may indicate some problems but they are not always a reliable indication of relative amplifier-to-amplifier "musicality."

4) Impedance: Solid state amplifiers generally produce less wattage into higher impedances (electrical resistances) than into numerically lower "loads." (Tube amplifiers often use output transformers which minimize this characteristic.)

   Ohm's Law tells us that a "perfect" amplifier would develop twice the wattage into a 4 ohm load than into an 8 ohm load. In the real world, however, low-impedance loudspeakers can drain an amplifier and cause the power supply rails to "sag," to deliver less operating voltage to the output stage. Thus an amplifier designed to produce 100 watts into 8 ohms may only be able to push 160 watts into 4 ohms rather than the 200 we would otherwise expect.

5) Number of Channels: In the golden days of stereo, this was never a problem. "All channels driven" simply meant "two channels driven." Easy. Unfortunately, in today's multi-channel world, things are murkier.

   The reasonable answer is that "all" still means the maximum number of separate channels on that amplifier's chassis. Unfortunately, that's usually the ***WRONG*** answer today where many amps with seemingly impressive specs look almost anemic when you realize that they develop the claimed "per channel" power rating only when 2 of the 5, 6, or 7 channels are actually working! Is this deceptive? Yep. Does it happen? All the time.

   The origin of this practice lies in early surround sound formats like Dolby Pro Logic, which couldn't deliver full-scale signals to all channels simultaneously. (That's just how matrixes work, by the way.) Amplifier manufacturers quickly noted this and reacted accordingly. "After all," they reasoned, "if this amp won't ever be asked to deliver full power to all channels at the same time, why should we rate it that way?" And so they began driving only two channels at a time, letting a power supply that might have been a heavy breather with five individual channels loaf along pushing just two. The result? Higher power ratings, of course!

But Dolby Digital, DTS, and discrete multi-channel sources like SACD and DVD-A altered that picture completely. Today's surround formats are totally capable of demanding full power to all channels simultaneously, so a multi-channel amp should deliver claimed power from all channels simultaneously. You'll need to read the specs closely to make sure.

A Question And A Surprising Answer

Let's check our understanding so far. Here are specs for two amplifiers. Can you tell which one would be the better choice?

Amplifier A: 100 watts RMS per channel, all channels driven from 80 Hz to 20 kHz into 6 ohms with no more than 1.0% THD.

Amplifier B: 75 watts RMS per channel, all channels driven from 20 Hz to 20 kHz into 8 ohms with no more than 0.5% THD.

Surprisingly enough, Amplifier B is probably better. Here's why:

1) Amplifier B's power measurements cover a wider power bandwidth (a full 20 Hz to 20 kHz) as opposed to amplifier A's more limited 80 Hz low-frequency limit.

   As we've already pointed out, reproducing low bass frequencies demands more from an amplifier. If Amplifier A specs power to only 80 Hz, that's a direct tip-off that something is lacking. In contrast, Amplifier B's specs go all the way to our 20 Hz low-frequency hearing limit.

2) Amplifier B outputs its claimed power into a higher load than Amplifier A, thus reducing its wattage.

   In fact, when you use Ohm's Law to figure things out, you'll find that Amplifier A and Amplifier B actually produce exactly the same power – if you measured Amplifier A's output at 8 ohms (a higher electrical resistance), you'd come up with 75 watts! Conversely, if you measured Amplifier B's output into a 6 ohm load, you'd get 100 watts! (That's one reason why we're opposed to 6 ohm ratings – they're deceptive!)

3) Amplifier B produces substantially *less* distortion than Amplifier A.

> Within limits, this is important. With half the distortion, Amplifier B is clearly operating in a more linear manner, at least under steady-state conditions. There may be several reasons for this and we can't tell exactly what they are from the specifications alone. The end result is that Amplifier B seems to impose far less of itself on the signal.

What's the point here? Simple: *DON'T LET THE POWER FIGURE ALONE CONVINCE YOU!* Power output measurements alone, as you now see, mean something only in context with other specifications. Look closely before you make up your mind.

Here's what you should look for:

Type of Power Rating:	Look for RMS power, not fictitious "peak power" ratings
Bandwidth:	This should cover the full 20 Hz – 20 kHz audible range. A "40 Hz – 20 kHz" rating is less desirable. An "80 Hz – 20 kHz" even more suspect. A "1 kHz-only" rating shouldn't be trusted at all.
Impedance:	Look for 8 ohm ratings. Those increasingly popular 6 ohm ratings are bogus and we don't think they serve any useful purpose. However, they *do* allow a manufacturer to make misleadingly higher power claims without telling you anything about an amplifier's ability to handle real low-impedance loads.
Distortion:	Within limits, lower numbers are better. In truth, however, anything below 1% is probably inaudible.

DIGITAL TECHNOLOGY

Introduction

Digital audio data appears almost everywhere today: CDs, DVDs, DAT tapes, DSS satellite broadcasts, personal computer sound files, speech synthesizers, even long distance telephone calls! But whatever the application, this digital data must pass through a digital-to-analog converter (often called a DAC, or D/A) before we can hear it as sound.

On one hand, the D/A converter's task seems simple: It translates binary numbers (a series of "0s" and "1s") into voltage pulses we eventually hear as audio waveforms.

However, the required speed and accuracy make this a critical task indeed. In fact, more than any other section in a digital audio device, the D/A converter determines how precisely the digitized signal will be restored to the analog domain.

In this section, we'll examine how a D/A converter operates, some of the difficulties involved in conversion, and some of the converter types currently used in digital audio products. In addition, we'll look at digital filters because they are so closely tied with converters that it sometimes becomes difficult to speak of one without discussing the other.

But before we do this, let's take a step back and look at how we convert an analog musical waveform to digital in the first place, how we move from the "analog domain" to the "digital domain." Once we've done this, you'll find CD players — and other digital source components — much easier to understand.

Basics

What is "analog," anyway?

This important question precedes almost everything else we'll discuss here. Although we won't get too technical, we need to begin with some elementary physics.

Sound waves are simply air molecules in motion. (Alright, alright, we can hear sounds through liquids and some solids, too, but not many of us regularly go to underwater concerts, do we?) Simply put, sound waves are caused when something (a sneeze, a bassoon, an airplane crash) momentarily squeezes air molecules together. Because these molecules are relatively anti-social — they naturally exist in what scientists call "a state of equilibrium" and resist being pushed closer to each other — they tend to spring apart immediately afterwards.

We describe this process as a series of ***compressions*** and ***rarefactions*** — in other words, a sound wave.

When we record music, a microphone converts these waves into voltages that are *analogous* to the continually changing density of air molecules. We describe these voltages as being in the *analog domain*. No, they aren't sound waves as such but these electrical voltages vary in much the same was as air molecule density does.

Digital is another matter entirely. There's nothing analogous about digital audio data — it bears no easily defined relationship whatsoever to an analog waveform. It is a series of binary numbers that only *represent* an analog wave. Thus, we say that digital audio data, although it can originate in the analog domain, inhabits the digital domain after analog-to-digital (or A/D) conversion.

In fact, from one point of view, digital audio data isn't even "real." It bears no relationship whatsoever to the sounds we hear but results from a totally arbitrary conversion process. Physicists sometimes refer to the digital domain as "*n-space,*" where *nothing* is grounded in our day-to-day reality. But it works – and that's the important part.

From Analog to Digital

The essence of digital audio is this: *an analog waveform can be represented by a series of binary numbers and A/D conversion accomplishes this process. However, we need to re-convert the numbers back into an analog waveform before we can hear it. That's called digital-to-analog (or D/A) conversion.*

Let's take a closer look.

There are two processes involved in converting sound into digits. The first is called **sampling**. The second, **quantization**. And both are comparatively simple to understand. Sampling simply means that we take a continuous event – a complex but continuous sound wave from an orchestra, for example – and cut it up into small "samples." Then we assign an arbitrary numerical value to each of these samples.

JUST A SILLY LOAF OF BREAD . . .

It may help to think of digitization this way: If you slice a loaf of bread, weigh each slice on a very accurate scale, and then note each slice's weight on paper, you've "digitized" that loaf of bread!

When you slice, you're "sampling" the whole loaf. When you weigh each slice, you're "quantizing" each slice. When you're done, you can add up the weights of each slice and determine how much the whole loaf used to weigh.

Of course, you'll end up with some inaccuracies. If your loaf was too big to fit into the slicer, you'd need to cut off a few inches. In electronic terms, the equivalent process would involve filtering out high-frequency content that the digital system couldn't handle. And that's exactly what an A/D converter's "anti-alias" filter does! If the blades in your slicer aren't spaced exactly the same distance from each other, you'll end up with uneven slices. (That's a crude description of "jitter," or timing inaccuracies in the initial sampling process.) If the blades aren't sharp enough, you may end up with some crumbs (other "conversion artifacts" we really don't want). If your scale isn't accurate enough, you won't be able to assign a precise value to each slice. (That's called "quantization error.") And if you can't weigh each slice accurately, you'll never be able to refigure the exact weight of the loaf before you sliced it in the first place!

In order to accurately capture sound waves, we have to "sample" them at a very high rate. In the case of a CD, we need to take 44,100 samples every second ***for each channel***. (That's why we say a CD's "sampling rate" is 44.1 kHz. It's just another way of saying 44,100 samples for every second of audio.)

The reason we need such a high sampling rate is because of the **Nyquist Theorem**, one of digital's basic building blocks. This tells us that the highest audio frequency a digital system can accurately convey is half its sampling frequency. Since we want to reproduce audio frequencies as high as 20 kHz, we need a sampling frequency of at least 40 kHz. The 44.1 kHz CD rate gives this with a little room to spare.

The bit structure needed to convey a high-fidelity audio signal must be up to the task, also.

Before we go any further here, let's define a few terms. A "bit" is simply the smallest unit of digital information: a digital "0" (current "off") or a digital "1" (current "on.") Bits combine in even-numbered lengths to form digital "words." Generally speaking, the more bits in a word, the more accurately that word can convey detailed information.

For example, CDs carry digital audio data in the form of 16-bit words or samples at the rate of 44,100 samples every second. Altogether, CD audio data streams off the disc at the rate of *1.41* ***million bits per second***!

As we've already said, a sample's accuracy is determined by the number of bits that make up each word. Specifically, an *n*-bit word yields 2^n possible quantization values. That's important because a digital system does not note each sample's value directly. Instead, the system uses a predetermined "look up table" based on bit structure. Rather than just computing the sample's real value and using *that*, the system selects the closest value already contained in the look-up table we just mentioned. From that perspective, we can say that the process of digitization is also one of approximation.

If that brings questions of sonic accuracy to your mind, you're perfectly justified. The answers to those questions, however, lie in the bit structure and sampling rate used by the digital system in question.

For example, an 8-bit word provides 2^8or 256 different quantization levels. CDs use 16-bit words to convey data which provide 2^{16} or 65,536 possible quantization levels in the look-up table.

Clearly, the more bits, the more accurate the sample. This directly translates into an output audio signal with lower noise and distortion.

A 16-bit sample is very accurate. Despite all the talk today about new "high-density" digital audio formats like DVD-A and SACD, properly implemented 16-bit performance is truly impressive. An analogy might help: If sheets of typing paper were stacked to a height of 22 feet, a single sheet of paper would represent one quantization level in a 16-bit system! If you think about how difficult it would be to see how much shorter that pile of paper would be with one sheet removed, you have some idea of 16-bit performance potential!

Digital Performance Parameters

Here's a review of common performance parameters in digital systems. We've already referred to some of these figures in the text and we'll refer to them again as we get deeper into digits. So here they are in tabular form.

Remember that higher numbers don't necessarily guarantee sound quality. A poorly implemented 24-bit system may not convey the musical delicacies of a carefully designed 16-bit system. Similarly, high sampling rates (96 kHz, for example) do not necessarily imply *audibly* better performance. High jitter, for example, may make the 96 kHz system less musical.

As we've already explained, "sampling rate" is important in that it determines the highest audio frequency that a system can process. As the Nyquist Theorem states, the highest audio frequency is one-half of the sampling rate. Practical implementation reduces this theoretical limit slightly. Thus, a digital telephone answering machine with a sampling frequency of, for example, 3 kHz will not process any signal higher than 1.5 kHz. CDs, with a sampling frequency of 44.1 kHz, will not convey any signal higher than 22.05 kHz in theory.

There's a difference between a system's basic sampling rate and its "oversampling" rate. A digital system's sampling rate is simply the number of samples coming from the source in one second. This sampling rate defines basic performance potential.

"Oversampling" (better called "resampling") simply refers to different ways of manipulating this data — usually by "speeding it up" — to improve sound quality. Many digital source components process data at an even multiple (2x, 4x, 8x, etc.) of the original sampling rate. Thus, a CD player may employ "8x oversampling" and thus have a re-sampling frequency of 352.8 kHz (44.1 kHz x 8) where a DAT machine operating at "8x oversampling" will have a re-sampling frequency of 384 kHz (48 kHz x 8).

Here are two tables that may help you understand this better:

Common sampling frequencies	*Application*	*Highest audio frequency* Theoretical/Practical
44.1 kHz	CD	22.05 kHz/20 kHz
48 kHz	DAT	24 kHz/22 kHz
96 kHz	DVD	48 kHz/44 kHz
192 kHz	DVD-A	96 kHz/88 kHz
2.8224 MHz	SACD	100 kHz/100 kHz [1]

[1] Remember that SACD's high-density data structure is single-bit delta/sigma rather than multi-bit PCM format specified for CDs and DVD-A discs.

System Bits	*Expressed As*	*Possible Quantization Levels* (2^n where n = number of bits)	*Theoretical S/N* (6 dB/bit)
1	2^1	2	6 dB
2	2^2	4	12 "
8	2^8	256	48 "
16	2^{16}	65,536[1]	96 "
20	2^{20}	1,048,576[2]	120 "
24	2^{24}	16,777,216[3]	144 "

Notes:

[1] "Red Book" CD standard

[2] Early digital studio mixers, editors, etc.

[3] Today's studio technology and part of the DVD-A standard

You'll see a lot of this information repeated in the DVD section. We've included it twice because it's important and having it in both places will save you a lot of page flipping.

ROTEL'S CRYSTAL BALL

But what about the new digital formats built around 24-bit (DVD-A, for example) or "single-bit" word structures and even higher sampling rates (SACD, for one)? Are they intrinsically superior to today's 16 bit/44.1 kHz CD standard?

In a word, the answer is "Yes."

But this doesn't mean these "high-density" standards will enhance our listening enjoyment! In short, while newer standards may be particularly attractive as "archival" formats (meant for long- term preservation of never-to-be-repeated performances), they may not present "quantum leap" audible benefits under most listening conditions. For example, a digital system built around a 24- bit word structure could provide a theoretical signal-to-noise (S/N) ratio of -144 dB; other circuit artifacts (power supply noise in particular) as well as ambient noise in our listening environments, would mask much of this theoretical improvement and would prevent us from hearing much, if any, difference. Despite some critics' long-standing tendency to "bash" today's 16 bit/ 44.1 kHz digital standard, the fact remains that it is superbly adequate — not to mention convenient — for almost all home music playback needs.

All that being said, we suggest that you read the chapter on DVD – in particular, the sections on DVD-A and SACD – before coming to any conclusions about these formats. You'll find a lot of information that will help you arrive at a more reasoned position.

From Digital to Analog

Digital Filtering

If you looked at digital data on a high-speed oscilloscope, you'd see . . . a very confusing picture indeed. Of course, if you already had your Ph.D. from MIT, you'd be able to interpret what you were looking at. However, if you're like the rest of us, we'll just ask you to accept the following.

The output of a D/A converter is a series of voltages which form a stepped – as opposed to smooth – analog waveform. The edges of these steps contain high-frequency energy that was not present in the original material when it was converted from the analog to digital domains.

To properly re-create the original waveform, these high frequencies must be removed with a low-pass analog output filter. First-generation CD players used so-called "brickwall" analog output filters to accomplish this. However, these filters were expensive, physically large, and added distortion to the signal.

In virtually all recent designs, these analog brickwall filters have been replaced by a combination of a digital filter (placed just in front of the D/A converter) and a more musically natural gentle-slope analog output filter. The digital filters "speed up" the converter and move residual "quantization noise" far away from the musical spectrum, thus allowing the use of less radical analog filters. These filters do not impose the phase shift and distortion characteristic of earlier brickwall designs. In addition to their musical merit, this combination of digital and gentle-slope analog filtration is actually less expensive to implement.

Essentially, digital filters are "dumb" computers that have been designed to do just one thing – speed up the operational speed of the D/A converter immediately following. They do this by giving the D/A converter more samples to deal with than contained by the source.

Contrary to common assumptions, digital filters do not usually change the value of source-direct samples. However, they do calculate values for additional "intermediate" samples that are inserted between the originals. Whether or not the value of these intermediate samples bears any relationship to the original samples is beside the point. (Some filters calculate related values; most do not.)

We use the term "oversampling" to describe this process. That's unfortunate, as the term is inaccurate. "Re-sampling" is a more accurate description of what the filter really does.

A "four times" or "4x" re- or over-sampling filter inserts three new samples between every original. An "8x" filter inserts seven new values. The result is that for every original sample we had from the source, we now have 4 (if the filter is a "4x" computer) or 8 (if we're using an "8x" filter.)

The real value of these new samples is that, in “speeding up” the digital processing, they also shift something called “quantization noise bands” further away from the music or soundtrack information we want to hear. With these noise bands further away from the audio data, we can then use a more gentle analog output filter (technically known as a “reconstruction filter”) after D/A conversion. These comparatively gentle analog filters are much more benign than their “steep slope” cousins used in first or second generation CD players. That’s because filters store and release energy over time. Steep slope filters in particular introduce significant amounts of phase shift and ringing into the original signal even as they blocked the quantization noise bands. That situation was, musically speaking at any rate, less than ideal.

In terms of performance, digital filters represent a great improvement over analog filters because a digital filter can be designed with considerable precision, whereas an analog filter design is inherently limited. Because digital filtering is a purely numerical process, the characteristics of digital filtering cannot change with temperature, age, etc. A digital filter might provide an in-band ripple of +0.00001 dB, with stopband attenuation of 120 dB. A digital filter can have almost perfectly linear phase response. In short, a good digital filter does not add appreciable noise or distortion to the audio signal. However, please remember that even the most advanced digital filter works in conjunction with the final analog output filter (a “reconstruction” filter) to bring you the sound you want to listen to.

D/A Conversion

The sheer number of bits involved shows the difficulties in accurately converting a digital audio signal back into the analog domain. D/A converters, found in every digital audio playback device, are the final and critical step in the digital signal chain. Let’s take a closer look at how one type, a PCM converter, operates.

> We need to make a quick distinction between two forms of digital data. The first is PCM (Pulse Code Modulation), the multi-bit format used for CD and DVD-A data. The second is delta-sigma (also called sigma-delta) or the single-bit data format used, for example, in SACD.
>
> Although we’ll get into details later, suffice it to say that every sample of a PCM data stream is “self contained.” Every sample contains all the amplitude information within its own structure.
>
> Single-bit data, on the other hand, is relative. Each sample’s level is only “equal to,” less than,” or “greater than” that of the previous sample.
>
> Complicating this issue is that it is possible to convert multi-bit PCM data to the analog domain using a so-called “single-bit” converter. (No, you do not need to lose any sleep over this.)

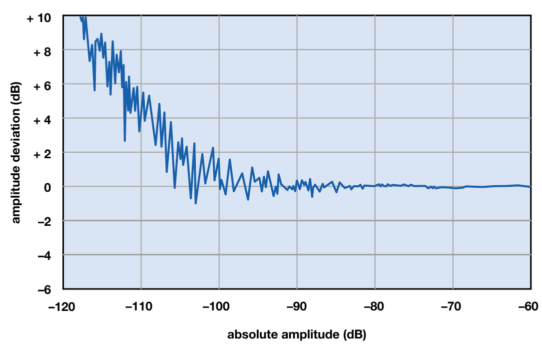
The reason we're mentioning this now is that most of the following information pertains to PCM (CD or DVD-A) data conversion.

A D/A converter IC (most handle at least two channels simultaneously, many now handle six or more) must accept this data stream from the source and output an analog voltage for every sample. In other words, an IC used in a CD player may output 88,200 discrete voltages every second while its multi-channel equivalent may generate more than 576,000! These voltage levels are filtered to create the analog waveform we eventually listen to through headphones or speakers.

A linearity test measures a converter's ability to output voltages at the proper amplitude. When a bit changes from 1 to 0 in a D/A converter, the analog output must decrease exactly by a proportional amount. Any nonlinearity results in distortion in the audio signal.

The amount that the analog voltage changes depends on which bit has changed. That's because every bit has a different impact on the final analog voltage. Technically, we explain this by saying that every digital word contains a series of binary digits, arranged in "powers of two." The first bit (the most significant bit or MSB) accounts for fully half of the analog signal's amplitude, while the last bit (the least significant bit or LSB) accounts for the least change.

There are several tests for D/A linearity. For example, the graph to the right shows a D/A converter's low-level linearity. Reproduced signals lower than -100 dB in amplitude most clearly show nonlinearity. For example, a -100 dB signal is reproduced at an amplitude of -99 dB, and a -110 dB signal is reproduced at approximately -105 dB. Depending on signal conditions, this kind of error may audibly alter low-amplitude information and effect the reproduction of ambience or "space" around the instruments.



This graph shows the linearity of a multi-bit D/A converter as signal level decreases. Note that performance is nearly perfect down to −90dB, below which the response becomes progressively less accurate.

Types of D/A Converters

Multi-Bit

All D/A converters accept digital words and convert them to an output analog voltage or current. However, various types of converters are used for audio applications. Multi-bit converters operate quite differently than low-bit (or "one-bit") converters.

Technically, a multi-bit D/A converter for either a CD or DVD player contains resistors and switches; there are two resistors per input bit. Each switch contributes an appropriately weighted current to the output. Specifically, the current that flows through each path is weighted by a binary power of two. If a current 1 flows from the reference voltage, I/2 flows through the first switch, I/4 through the second switch, I/8 through the third switch, etc.

The digital input bits are used to control the switches that produce the analog output; a 1 value closes the switch to contribute a current, while a 0 value leaves the switch open, contributing nothing. We can see that a 1 in the MSB will produce a larger current (and hence a larger output voltage) than a 1 in the LSB. Similarly, each combination of input bits results in a unique voltage output.

THE DIGITAL DERBY

Imagine, if you will, the beginning of the Digital Derby, a race with 16 horses in the field. (Yes, we're talking CD here – if we were referring to DVD, we'd need a 24-horse analogy.) Before the race begins, a horse enters each starting stall sequentially. When all 16 horses are in, the gates for each stall lift simultaneously and all the horses gallop off at the same time.

That's very similar to what happens in a multi-bit converter. The 16 bits that make up each digital "word" file sequentially into the separate registers of a D/A converter. The converter "weighs" each one, calculates the proper amplitude of a corresponding analog signal and outputs that combined analog voltage pulse.

Why, then, did some manufacturers introduce 18-, 20-, or 24-bit converters to provide greater playback fidelity for 16-bit recordings? The rationale for this lies in flaws inherent in D/A converters. Except in theory, 16-bit converters cannot fully convert a 16-bit signal without a degree of error. In order to realize full sound potential, the conversion must have a greater dynamic range than the final recording.

High-bit converters, for example, usually provide better conversion of CDs' 16-bit words than do their 16-bit cousins. When done correctly, higher-bit conversion improves amplitude resolution by ensuring linear conversion of the 16-bit signal. A 20-bit D/A converter has 1,048,576 levels, almost 16 times as many output levels as a 16-bit converter. A 24-bit converter, as you've already seen, provides 16,777,216 output levels. Any nonlinearity is correspondingly smaller, and increasing the converter's wordlength results in an increase in S/N ratio.

However, the measured linearity of a multi-bit D/A converter, and not the number of bits it employs, ultimately determines its accuracy. For example, a highly nonlinear 20-bit converter may be inferior to an accurate 16-bit converter.

The Multi-bit Dilemma

Multi-bit conversion represents a classic approach but can be problematic. Because a large number of bits are used to form the representation, and because each bit may have an error unequal to the others, the overall error varies with each sample, and is thus difficult to correct.

Moreover, any instability in the current sources will yield distortion at the output. In addition, a problem known as zero-cross distortion can cause a switching glitch in the output waveform each time the waveform crosses the center line. The error is particularly troublesome when reproducing low-level signals, because the fixed-amplitude glitch is proportionally large with respect to the signal. In practice, calibration procedures during manufacture, and sophisticated circuit design are required to achieve high performance and maintain it over the life of the converter. Understandably, manufacturers have sought to develop alternative conversion methods, including low-bit converters.

The Low- (or Single-) Bit Answer

Limitations in multi-bit converter architectures have stimulated development of low-bit converters. These systems are characterized by very high oversampling rates, noise shaping, and wordlengths of one to (generally) five bits.

An engineer might describe a multi-bit converter as a high-resolution quantizer operating at a low sampling rate. That same engineer would then describe a low-bit converter as a low-resolution quantizer operating at a high sampling rate.

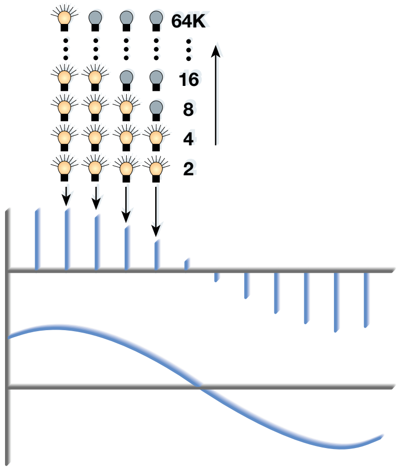
Although it is not easy to see how one bit (or a few) can replace 16 or more bits, consider another analogy, this one a bit different from the Digital Derby described above.

Multi-bit converters are like a row of light bulbs, each connected to a switch. Sixteen bulbs, for example, each with a different brightness, can be lighted in various combinations to achieve 2^{16}, or 65,536, different brightness levels.

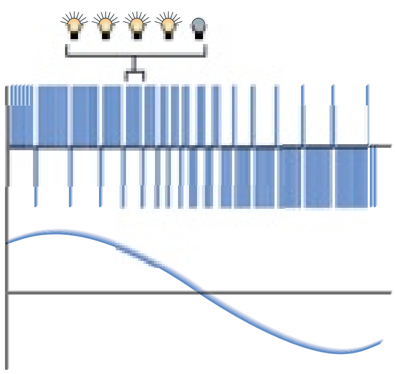
However, relative differences in individual bulb intensities will introduce error into the system. A certain switch combination will not always produce the desired room brightness. Similarly, multi-bit converters introduce error as they attempt to reproduce the audio signal. Low-bit conversion uses a wholly different approach. Here, signal amplitude (the equivalent of room brightness) varies according to how many times a second a single switch (controlling the equivalent of one light bulb) is toggled on and off.

So, instead of many bulbs and switches, we're now using only one bulb and one switch. Dynamically, if the bulb is switched on and off equally, the room is at half brightness. If the bulb's on-time is increased, room brightness will increase.

Similarly, low-bit converters use one bit to represent an audio amplitude, with very fast switching and very accurate timing. Low-bit technology is an inherently more precise method of representing an audio waveform. Conventional multi-bit conversion divides the signal in multiple amplitude steps. Low-bit conversion divides the signal in time, keeping amplitude changes constant and this high or low level pulse represents an analog audio signal.



In Pulse Code Modulation (multi-bit) processing, the analog sine wave is represented by a series of digital "words" which vary in amplitude depending on which combination of 16 bits is turned on. This type of digital processing can be compared to lighting a room with 16 bulbs, with each succeeding bulb having twice the brightness of the previous one. The overall brightness is determined by how many bulbs are lit.



In Pulse Density Modulation (or one-bit processing), just one digital value switching on and off very quickly represents the analog signal. This type of processing would be similar to lighting a room with a single flashing light, with the overall brightness determined by how long the light stays on and how long it is off.

For example, the previous diagram shows how a single constant-width pulse, with either a high or low level, can reconstruct a waveform. Alternatively, a few bits can be used to reconstruct the output signal; these are called low-bit converters because the signal is typically quantized to 5 or fewer bits. Low-bit D/A converters use sophisticated processing to implement noise shaping, and decrease the high in-band noise levels otherwise present in low-bit conversion. A variety of D/A architectures have been devised, with different algorithms and orders of noise shaping. A true 1-bit system outputs a binary waveform – albeit at a very high rate – to represent the audio waveform. In practice, 1-bit systems do not always perform well and are not commonly used. Practical low-bit systems output a low multi-bit signal, and use noise shaping algorithms to reduce audio-band noise.

A Balanced Perspective

Both multi-bit and low-bit conversion have merit and a high-quality digital audio player can use either provided, of course, that they are well-designed examples of their genres. However, high- quality multi-bit converters are more expensive than low-bit converters.

Finally, it is important to remember that although choice of the D/A converter is important, ultimate performance can depend on the quality of the circuits surrounding the converter. For example, poor power supplies, improper grounding and shielding, a below-par analog stage, and other weaknesses can undermine performance of even the best converter IC.

Compact Disc Laser Pickups

Introduction

CD and DVD players recover and process data encoded on discs in the form of a single spiral of tiny pits (or bumps – it all depends on how you look at them!). The recovery process begins with the laser pickup used to read disc data. Of course, to do this efficiently, the laser needs an automatic tracking and focusing system in addition to the reading capability.

Among other technicalities, early CD players used to be distinguished by the number of "beams" generated by the laser system – three or one.

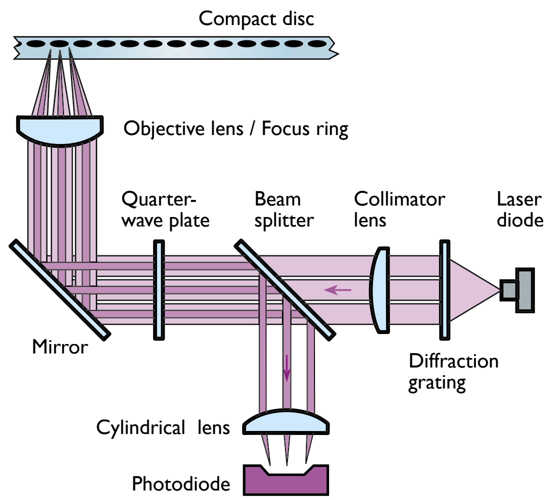
We'll consider the more common three-beam design first but, before we do, want to make an important point here: As a practical matter, the differences between triple-beam and single-beam lasers really don't matter today!. That's because the strengths of each (faster access for triple-beam systems, better ability to read damaged disc areas for single-beam systems) are minimal now.

As far as DVD players go, there is no issue here at all – almost all DVD optical systems are built around the "single-beam" approach since the track pitch is so much smaller for DVD data. As you'll soon see, single-beam pickups use a technique called Differential Phase Detection that keeps the laser system focused on the pit spiral by comparing the phase of the four photo detectors employed. (Don't worry if you don't understand this right now. Read on …) However, you should know that some DVD players also use a separate three-beam laser optical system to read CD data.

Three-Beam Pickups

A three-beam optical pickup contains a laser diode, diffraction grating, polarization beam splitter, quarter-wave plate, and several lens systems, as shown in this diagram.

Notice that the light from the laser point source passes through a diffraction grating. (This is a screen with slits spaced only a few laser wavelengths apart.) Often called a "beam splitter," this diffraction grating creates three beams from the original one.



The center beam is used for reading data and focusing. The two secondary beams aid in tracking the disc's pit spiral. Beam focus is critical as an unfocused condition may result in inaccurate data. The three-beam laser uses an optical property called astigmatism for auto-focus. A cylindrical lens detects out-of-focus conditions. As the distance between the objective lens and the disc reflective surface varies, the focal point of the system changes, and the image projected by the cylindrical lens changes its shape. A photodiode reads this change and generates a focus-correction signal when needed.

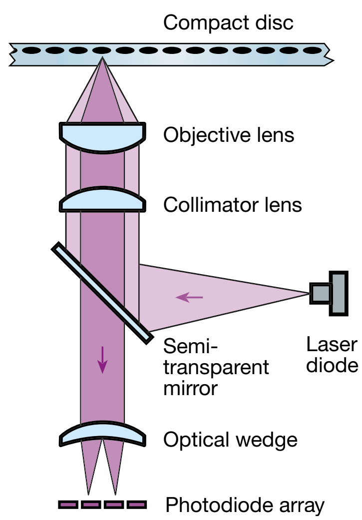
As we've already said, the laser's single beam is split by a diffraction grating. This creates two secondary beams that reach the disc's surface with the central beam. The central beam spot covers the pit track, while the two tracking beams are aligned above and below and offset to either side of the center beam. When the main beam mistracks, the difference signal generated by two side beams corrects the problem.

The laser assembly automatically generates this difference signal when the whole assembly is slightly out of alignment – where one of the side beams reads more pits than land area and is therefore weaker than the other side beam which is simultaneously reading more land area than pits.

An easy way to think of this whole arrangement is consider the side beams as "bicycle training wheels" for the main beam. While that image isn't very romantic – or scientific, either – it'll do for now!

Single-Beam Pickups

The optical components of a one-beam pickup are shown here. Notice the different photodiode array that's used to read data as well as generate tracking and focusing signals.



A semi-transparent mirror directs light from the laser diode to the disc surface. Light reflected from the disc passes back through the mirror and is directed through a wedge lens. The wedge lens splits the beam into two beams that strike an array of four horizontally arranged photodiodes. This array is the Differential Phase Detection circuit we mentioned earlier. Their summed output provides the data signal which is then demodulated to yield both audio data and control signals for the laser servo system.

Both three-beam and one-beam pickups can achieve excellent performance. As with any mechanical and optical devices, performance is ultimately defined by the quality of engineering design, selection of components, and manufacturing techniques. For further reading, we strongly suggest Ken Pohlmann's ***Principles of Digital Audio***, McGraw-Hill.

HDCD®: STRETCHING THE DIGITAL ENVELOPE FOR HIGH-QUALITY SOUND

Introduction

We're at an important transition in digital audio today. Not only are we moving towards wider acceptance of multi-channel sound in place of conventional two-channel "stereo," but we're also well on the way towards more wide-scale acceptance of advanced audio technologies like DVD-A and SACD that offer some potentially important improvements compared to today's CD-standard 16 bit-word length sampled at 44.1 kHz.

The multi-channel vs. stereo debate is somewhat inevitable. Although it began back in the early '70s when "quad" sound first appeared, it is now more germane than ever because it is fueled by the new surround technologies originally developed for movie soundtrack reproduction. However, as we've tackled this issue head-on in another section of this book, we won't repeat ourselves here. However, a few brief introductory comments are appropriate.

Some engineers, designers, and audiophiles feel that the CD standard is not sufficient for the best sound reproduction and point to DVD-based storage media as the best way to achieve "studio quality" sound in the home

Despite the rapid proliferation of DVD players (one resides, if you believe the latest surveys, in one out of every three homes in the U.S.), the CD still reigns supreme as the commercial software format that actually carries music into our homes. (The word "commercial" was specifically placed in the last sentence to preclude someone yelling "Internet, Internet!")

The fact is that it will take several years until the number of DVD-based releases comes close to the number of CD titles ready and waiting today for you to give them a home. This argues that there's a potentially huge market for better CDs. And better CD players. That's where HDCD® (High Definition Compatible Digital) comes in.

Even with that as an introduction, rest assured that there's now a lot more to this picture. Pacific Microsonics, the company that brought HDCD to the audio world, was recently purchased by Microsoft. Obviously, this "800-pound gorilla" didn't buy Pacific Micro just to participate in the (comparatively) tiny revenue stream generated by royalties on the HDCD process. The behind-the-scenes activity now going on will probably make HDCD even more important in the future.

Before we polish up our crystal ball, however, let's get some of the basics in place.

CDs Aren't Perfect?

Most of the objections to CDs are rooted in some basic limitations. According to standards initially formalized by Philips and Sony (and formalized in what's known as the "Red Book"), CDs carry data in the form of 16-bit words at the rate of 44,100 words per second per channel. No less. And no more.

In real-world terms, these Red Book standards mean a CD's high-frequency response is limited to approximately 20 kHz and that the theoretical noise floor is, at best, 96 dB below full output.

The reasons for these limitations are technically complex. Rather than delve into them in detail, we'll make three quick points now:

1) The Nyquist Theorem states that the highest frequency a digital system can reliably process is half the sampling frequency. As a CD's sampling frequency is 44.1 kHz, the highest audio frequency it can carry is 22.05 kHz. In practice, high frequencies are commonly limited to 20 kHz and few discs have useful response anywhere near this figure.
2) Each bit in a digital system is equivalent to a 6 dB improvement in potential signal-to-noise performance. An 8-bit system "maxes out" at -48 dB (8 bits x 6 dB each) while a 16-bit system can reach -96 dB performance levels. Again, in practice, this ideal spec rarely sees the light of day (or laser).
3) All digital systems generate "quantization errors" which result from differences between actual sample values and the numbers a digitizing system picks to represent them. With CDs, each infinitely variable sample is "squeezed" into one of 65,536 predefined "cubbyholes." The fit is never exact and the differences manifest themselves as unwanted low-level spurious noise components added to the original signal.

Sonic Implications

Although they admired the CD's convenience, critics disagreed about the sonic performance – particularly of early players. "Brick wall" analog filters at 20 kHz, they said, destroyed the ambience or sense of space around the musical signal. In addition, the 16-bit limitation didn't allow for accurate reproduction of the very small-scale differences (called microdynamics) that characterize live music. And quantization errors often masked low-level harmonic structures that allowed a knowledgeable listener, for example, to distinguish a mahogany-backed Martin D-18 guitar from a rosewood D-35.

Professionally Speaking

The professional audio community also recognized the CD's somewhat restricted archival qualities and gradually switched from 16- to 20- and, later, 24-bit digital formats for recording, editing, and mastering. The reason was simple – the "higher bit" systems sounded better and gave the production crews much more data to work with. A 20-bit system, for example, with four more bits of data, sports a signal-to-noise improvement from -96 to -120 dB. A 24-bit system, at least in theory, boasts an even more impressive -144 dB spec.

These new formats also allowed higher sampling rates which, in turn, resulted in extended-high frequency response. In addition, quantization errors were also substantially reduced.

The Heart of the Matter

But a substantial problem still remained – even though new digital formats delivered greater fidelity, the "Red Book" standard remained. That meant that every CD player and, more importantly, ***every*** *CD*, had to adhere to what some perceive as a very limiting standard indeed.

HDCD's Beginnings

One of the more vocal early critics of CD sound was Keith Johnson, an engineer for San Francisco's Reference Recordings. One of that company's first all-digital productions occurred in 1984 and resulted in the release of Berlioz's "*Symphony Fantastique*" (RR-11).

During this and later sessions, Johnson's thoughts on digital deficiencies and possible correction strategies began to crystallize. Michael ("Pflash") Pflaumer, an engineer with extensive experience in software design and microcomputer applications, joined Johnson to explore possible solutions.

Their eight-year research project began with an examination of how human hearing works, especially how the inner ear's basilar membrane and hair cell tips respond to different types, or "envelopes," of musical energy.

The highly complex findings convinced Johnson and Pflaumer that the best way to overcome a CD's aural limitations would be, in their words, "a conjugate digital encode/decode system." In English, this meant a "double-ended" system in which an information-rich digital signal is first encoded onto a CD and then decoded during playback.

Johnson and Pflaumer also rethought the problems inherent in converting an analog signal (a mic feed at a live recording, for example) to the digital domain and back again for reproduction. They examined the effects of anti-aliasing filters (the filters used to block extraneous high-frequency sounds from interfering with A/D conversion), jitter (timing errors in the converter itself), passband ripple, and many other performance aspects.

Based on their research, Johnson and Pflaumer set rather unusual standards for dynamic range, signal-to-noise ratio, distortion, and frequency response. The standards were unusual in that they were precisely that – *dynamic*. They changed depending on the nature of the music and how the ear responded to different signal envelopes.

Their double-ended (encode/decode) strategy allowed some interesting approaches. Among them was the decision *not* to make encoding and decoding equivalent functions. Indeed, Johnson and Pflaumer allocated most of the computational horsepower to the *encode* process. This included real time analysis of the incoming signal, on-the-fly selection of the appropriate processing algorithms, and creation of an instruction set to let the decode circuitry know exactly how to reconstruct the original signal.

HDCD Encoding

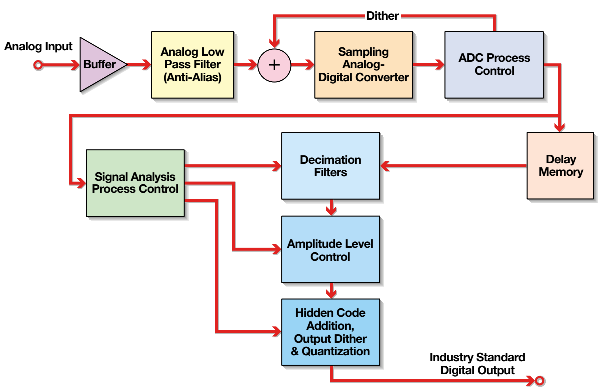
In practical terms, the HDCD encode process begins with a high-resolution signal, either analog or high-density digital. If the original signal is analog, the Model Two HDCD Converter/Encoder first converts it to a 24-bit signal at either 176.4 or 192 kHz. Then it buffers the signal to delay it just long enough to determine which sonic aspects would be most compromised when converting to the CD-standard 16-bit/44.1 kHz PCM format.

That analysis (and the resulting reformatting) benefits from 400 MIPS (millions of instructions per second) of computing power supplied by a bank of 8 Motorola DSP56009 microprocessors. These "number crunchers" run custom software programs to identify various distortions common to digital processing. The distortions fall into two general types: (1) "Additive" or those artifacts added to the original signal by both A/D conversion and subsequent D/A conversion and (2) "Subtractive" or actual data losses that result when the enhanced data stream is reconfigured to meet the CD format's 16-bit, 44.1 kHz limitations.

In general, HDCD encoding follows this process:

The high-density digital signal is first ***decimated*** (digitally reduced to a lower sampling rate and then to the final 44.1 kHz CD-standard sampling rate) by a set of variable slope filters continuously chosen to optimize moment-to-moment program content. (Pacific Micro calls this *Dynamic Decimation*, by the way.)

There are several advantages to dynamically optimizing the anti-alias filter's "transition band response" for faster settling times around transient edges and minimal response ripples when the signal contains rapidly changing high-frequency content. For starters, this improves resolution and enhances timbral accuracy. In effect, the system selects different anti-alias filters depending on the content of the program material to provide the optimum one for each condition. This dynamic filter switching technique is unique to the HDCD process.



HDCD Encoding process – For professional applications and to make sure the HDCD process is easily integrated into recording and mastering studios, the Model Two encoder makes the signal available in 24-bit form. Once fully approved by artists and producers, the edited signal is run through the Model Two again for final encoding and reduction to 16 bits.

Even after final decimation, the 44.1 kHz, 24-bit signal contains far too much resolution and dynamic range to fit onto a 16-bit CD, so the signal next undergoes ***dynamic range modification*** through a combination of soft peak-limiting and average-level compression. To analyze and determine effective yet musically neutral signal processing strategies, the Model Two looks at average mid-band energy, both peak and average high-frequency energy, and overall signal level.

The first, called "**Peak Extension**," is intended for infrequent, high-intensity peaks. It is a switchable, ***instantaneous*** soft limit which benefits from one-to-one mapping so that all reduced peaks can be restored in the decoder.

The second, "**Low-Level Extension**," is optimized for the other dynamic extreme – very weak signals. Here, the system uses compression, which very gradually raises the gain when the average level drops below a threshold.

The HDCD system determines gain based on an average of the broad middle frequencies in the signal – highs and lows are attenuated. The main signal has only its gain modified. Frequency response is unaltered and this minimizes any audible artifacts encountered in un-decoded playback.

High-frequency dither – optimized to reduce distortion and enhance resolution – is then added when the signal is re-quantized from the Model Two's nominal 24-bit resolution to the final 16-bit CD word length. (Johnson and Pflaumer estimate that the final decoded information exceeds 19- bit resolution even though the format is still 16-bit.)

> Remember that all CDs, even those labeled 20- or 24-bit, still use 16 bits of information per digital "word." The 20- or 24-bit label refers only to the resolution of the master tape used to create the 16 bit-CD. That's one of

the reasons why we consider HDCD such an important development. It offers the consumer greater than 16 bits of resolution from the Red Book CD format.

Note that these parameters are not fixed: The recording or mastering engineer can select a combination of process parameters precisely suited to the program material. This means that the tradeoff between performance when decoded and artifacts when not decoded is in the hands of the engineer!

In other words, ***HDCD encoding is variable***. Consequently, ***HDCD decoding is also variable*** as the decoder simply executes the command sequences embedded in the disc data. (See ***Getting to the Decoder*** below.)

Also note that HDCD encoding applies mostly to dynamic or frequency extremes. Although HDCD encoding increases the average signal modulation, the process itself is designed to have minimum impact on musical integrity. That's why the HDCD strategy remains *compatible* – i.e., why an HDCD-encoded disc can be played back on a non-HDCD player.

Getting to the Decoder

Conveying the instruction set to the HDCD decoder is a challenge. After all, the only link between the encode process and decoding is the CD itself and "Red Book" standards cover almost every imaginable method of conveying data. That's why the command stream, the data that tells the decoder what to do, had to be hidden or encrypted in some way so that it didn't add any artifacts of its own to the audio data.

The research team hit on the idea of using the Least Significant Bit (LSB) of the CD's 16-bit digital word as a carrier for the encrypted signal. Infrequently used to carry musical information, the LSB (the last bit in the 16-bit chain) was ideal. Even so, Johnson and Pflaumer took care to use only 1% to 5% of the available LSBs to carry decoder commands. This means that 15.95 to 15.99 bits of the CD's 16-bit format are available for musical information. For all intents and purposes, this is sonically identical to 16 bits.

Because of this approach, HDCD-encoded discs are totally compatible with conventional CD players and processors. When the HDCD filter/decoder senses the proper LSB sequence, it processes the disc data according to the continual series of instructions carried in subsequent LSB sequences. The decoding process is considerably simpler than the encoder's extensive data analysis and reformatting and accounts for somewhat less than 40% of the total number crunching involved.

The relatively "dumb" decoder IC simply executes various combinations of self-contained data reconstruction strategies when triggered by the appropriate LSB sequences.

Summary of HDCD commands carried in the "hidden" LSB sequence:

1) the filter choices used for decimation to the final sampling frequency
2) whether the Peak Extend algorithm is On or Off
3) the nominal values of the gain for Peak Extend and Low-Level Extension algorithms.

At The Decoder

Any component carrying the HDCD logo has at least one common element –a decoder/digital filter IC developed to process HDCD-encoded audio data. The first of these was Pacific Microsonics' PMD-100. It, the current PMD-200, and 13 other ICs by 11 different manufacturers, replaces the digital filter found in a number of products ranging from CD players, to advanced DVD-A players. Although the original PMD-100 was "pin compatible" with the most popular digital filters of its day, newer ICs don't follow any particular archetype.

Whatever the origin, an HDCD-capable decoder/filter IC is complex although, of course, some are intended for more rigorous applications than others. Microsoft's PMD-200, for example, is built around a Motorola processing core and offers bit lengths of 16, 18, 20, and 24 at sampling rates of 44.1, 88.2, 96, 176.2, and 192 kHz. However, it doesn't do Dolby Digital, DTS, or MPEG audio decoding as Zoran's Vaddis III and IV ICs can.

The important thing here isn't a comparison of IC capabilities, it's that many IC manufacturers see the merit of HDCD processing and provide multiple solutions so that hardware manufacturers like Rotel can pick the best decoder/filter for their particular application.

Regardless of which decoder/filter IC we're talking about, they do many of the same things, even if they do them in slightly different ways.

First, these ICs **decode an HDCD-encoded digital data stream** from a CD or other source. (More details on how this happens in just a few moments …)

For both HDCD and non-HDCD sources, these ICs **operate as a sophisticated oversampling digital filter**. In fact, many of them also **linearize and otherwise optimize the digital to analog converters** immediately "downstream" in the signal path. This avoids many of the glitches commonly associated with conventional digital to analog conversion.

An important implication is that the advanced filtration **optimizes playback of even conventional (non-HDCD) discs.** This is due, in part, to the inherent performance characteristics of all HDCD decoder/filters. For example, ripple, a measure of frequency response variations within a filter's so-called "passband," is a minuscule 0.0001 dB from DC to 20 kHz. And "stopband attenuation," or how well it suppresses unwanted frequencies, must be an impressive -120 dB.

Another potential advantage (depending, of course, on the exact IC chosen by a manufacturer) is ability to supply variable ultrasonic dither (random digital noise anywhere from 30 to 80 kHz in frequency) to optimize D/A linearity. System designers and circuit engineers select the dither type and level to precisely compliment the performance parameters of the D/A converters in a particular CD player.

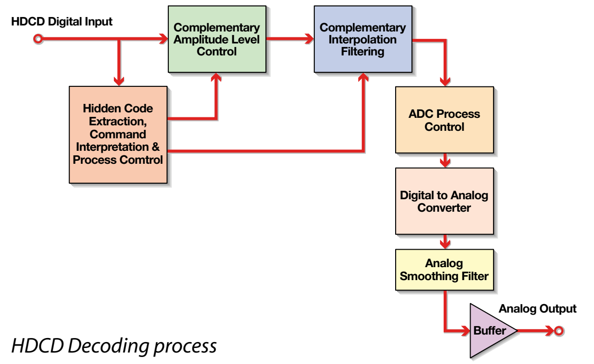
For instance, a designer can minimize quantizing errors by selecting one of six levels of high- frequency weighted dither to correct quantizing errors in multibit D/As or "white triangular" dither if they've chosen single-bit D/A ICs.

Other corrective circuitry includes "deglitching" and slew rate adjustments to better match the filter/decoder to the chosen D/A ICs.

Other variables include selections for both input and output bit lengths and oversampling rates. Some ICs accept signals from 16 to 24 bits long at sampling rates from 44.1 to 192 kHz and outputs16- to 24-bit words at 2x, 4x, or 8x the input sampling frequency.

Even though it's little used today, most ICs provide de-emphasis (a form of EQ derived from the old RIAA-style high-frequency curves needed to get a balanced spectrum on and off an LP music "digital EQ") in the digital domain.

Although HDCD-capable ICs are not difficult to include in components today, engineers need to observe a few cautions. First, there can't be any DSP (digital signal processing: phase inversion, surround format decoding, etc.) *prior* to the HDCD decoding as any processing would obscure or strip out the LSB-embedded command codes and render HDCD decoding impossible.



HDCD Decoding process

Second, an HDCD-decode circuit must include "gain scaling" to compensate for the "Peak Extension" option available during HDCD-encoding. "Peak Extension," as we've already described, reduces overall signal level by 6 dB during encoding to allow large-scale peaks to pass onto the disc without exceeding whatever dynamic range constraints a particular format is subject to. The combination of "Peak Extension" and gain scaling adds 6 dB to the decoded peak digital signal level, which, in turn, increases a recording's dynamic range by 6 dB. (This is, obviously, more important with CDs than with the DVD-A format, for example, as DVD-A's dynamic range performance is more dependent on its 24-bit data structure than anything else.)

HDCD recognizes that the peak digital signal level prior to decoding is already 0 dBfs (digital full scale) and cannot be increased. So, HDCD ICs *decrease* the average level of Peak Extend recordings by 6 dB to make room for the increased peak signal level.

Although this digital compansion (***com***pression during encode, ex***pansion*** during decode) solves a thorny problem, it creates another, this one psychoacoustic in nature.

Our perception of "loudness" is determined by a signal's average level, *not* peak levels. For that reason, decoded "Peak Extension" recordings will sound 6 dB quieter than non-Peak Extension recordings. This perceived level shift needs to be corrected. Gain Scaling during decode automatically sets the playback level of an encoded Peak Extension source 6 dB higher than a non-Peak Extension source.

Gain Scaling is thus user-transparent in that HDCD decoder/filters automatically engage the process whenever it senses the "Peak Extension On" command in the disc's LSB sequence.

Gain Scaling eliminates the otherwise unavoidable inconvenience of adjusting playback levels to maintain the same perceived loudness every time a customer replaced a Peak Extension HDCD-encoded disc with any non-Peak Extension disc (whether HDCD-encoded *or* conventional!)

This process can take place in either the digital domain (within the HDCD IC itself) or in analog circuitry after D/A conversion. Although digital gain scaling is easier, it comes at the price of 1 bit of resolution (6 dB = 1 bit). For that reason, Gain Scaling, particularly when used for CDs, should be performed with glitch-free, highheadroom analog circuits triggered by a detector in the IC. These analog circuits, of course, must be immune to any possible overload when analog signal levels from the D/A converter increase by 6 dB.

Going Forward . . .

A few years ago, one writer noted "If HDCD fails to attract commitment from at least one major record label or is simply outgunned in the marketplace by organizations with competing enhancement strategies, it will remain an esoteric footnote to the history of the CD."

Well, HDCD is far more than a footnote today.

Almost all of the major record labels have committed to HDCD. More than 5,000 HDCD recordings accounting for over 300 million discs now in consumer hands have been made by leading mastering studios around the world.

Of these, more than 250 have appeared on the Billboard Top 200 chart, and more than 175 have been nominated for GRAMMY Awards. These releases cover all musical tastes from mainstream pop to truly esoteric. That's a far cry from late 1992, when Reference Recordings released the first (and, at that time, the ***only***) commercially available HDCD-encoded disc ***Testament*** (RR-49), featuring the Turtle Creek Chorale with the Dallas Wind Symphony.

. . . To The Future

The CD

Although some may think that new proposals built around DVD's extended data capacity (24-bit words at sampling frequencies up to 192 kHz) may eclipse HDCD's ultimate potential, we see some problems with this view. For one thing, it ignores the huge base of consumers now familiar and quite comfortable with the CD. Remember that an HDCD-equipped player will also improve the fidelity of the entire library of CD music available today. That's an enormous reservoir that any new format will take at least a decade to equal.

Remember also that *HDCD is a* ***"non-obsoleting"*** *technology*. It doesn't ask the consumer to write off the often-considerable investment he or she has already made in software. A CD player with HDCD capability is an excellent candidate to replace a customer's old player and allows more musical enjoyment from CDs already owned. In addition, the decoder/filter's "user-transparent" nature makes HDCD playback a "no-brainer." The only difference a consumer will experience between an HDCD-encoded and a non-encoded disc is the front panel LED that indicates HDCD decoding.

Another Perspective

HDCD provides an incentive unmatched by other CD enhancements — the opportunity to *buy* or *sell* something. HDCD-equipped components give both consumers and retailers the opportunity to add performance capability to an otherwise satisfactory system.

For retail salespeople, we suggest demonstrating an HDCD-capable component with a conventional CD. Once you've demonstrated HDCD's effectiveness on material your customer already knows, you've overcome a major sales obstacle. If you follow that presentation with a short explanation of HDCD's full benefits and then play an appropriate HDCD selection, the benefits will probably be obvious to even jaded audiophiles. Obviously, pre-qualifying your customers will save you a great deal of time and effort.

Compatibility Issues

There's still some confusion out there regarding player and disc compatibility. Here's a wrap-up you might find helpful.

1) ***Non-HDCD disc, no HDCD decoder:*** No difference from what you've been listening to for the last seventeen+ years. Some players (and discs) are very good; some are, well, not so good.

2) ***Non-HDCD disc, HDCD filter/decoder:*** As explained above, the filter functions of all HDCD decoder filter ICs use advanced algorithms that should prove very effective in bringing out details less capable digital filters lose.

   Note: We're objective enough to realize that this claim is difficult to verify in an A/B or A/B/X comparison. However, our subjective evaluations seem to verify the performance advantages.

3) ***HDCD-encoded disc, HDCD filter/decoder:*** HDCD advantages include superior reproduction of instrumental and vocal timbre, a more extended and solid soundstage, better ability to follow individual voices or instruments in complex musical pieces, almost total transparency — even when directly compared to high-speed analog master tapes — and absence of the harsh artifacts some claim intrinsic to less sophisticated digital processing techniques.

One early claim for the HDCD process was that an HDCD-encoded disc sounded better than a conventional disc even when played back with *no* HDCD decoding. These claims apparently originated from the fact that most (approximately 60%) of HDCD's number crunching takes place during encoding and that decoding is a far less complex process. Some reasoned that non-decoded playback would therefore give you 60% of the "benefits."

We don't agree.

Remember that most of the encoding process involves real-time analysis of an incoming signal and the formulation of strategies to preserve audibly relevant information as the data stream is reduced to the "Red Book" CD standard. Part of the program guiding the encoding process determines what portions of the extended bit signal are essential for accurate playback and which signal elements can be safely "lost" during reformatting. By definition, this is "data compression."

As you've already seen, HDCD employs several proprietary strategies to insure that all audibly relevant information is preserved. Keys to these strategic decisions are embedded in the reformatted 16-bit/44.1 kHz data stream used to master HDCD-encoded CDs.

With these things in mind, it should be no wonder that we strongly recommend that all HDCD encoded discs be played back through an HDCD decoder. This insures that data reconstruction is complete and that you'll hear everything the artist and producer intended you to hear. Anything less just isn't good enough.

Beyond the CD

Perhaps the only thing we need mention here is the phrase "A Windows Media Technology." That's the tag line you'll find on the HDCD web site. As we've already mentioned, Pacific Microsonics was recently acquired by Microsoft. We'll certainly see more in the future as Microsoft incorporates various aspects of HDCD technology to support its PC and web developments.

On the other hand, we'd be less than complete if we didn't mention (again) that the HDCD process is almost as applicable to the DVD-A format as it is to the CD. For those of you concerned that HDCD and MLP (Meridian Lossless Packing – the data compression algorithm used on some DVD-A discs) may be incompatible, rest easy.

Remember that the "L" in MLP means "Lossless." By definition, then, the HDCD data stream will be perfectly preserved through the MLP "packing" and "unpacking" process. In fact, the official MLP spec includes provision for an "HDCD bit." In other words, MLP provides detection of HDCD signals in the packed MLP data, thus allowing appropriate HDCD decoder/filter operation. .

There are similar provisions in the Dolby Digital and DTS specs, too. Both of these formats allow detection of the "HDCD bits" that trigger proper decoding.

The only potential difficulty may arise from taking advantage of DVD-A's automated downmix" process (where six channels can be automatically reduced to two for stereo playback). In inexpensive players, this process can occur before there is an opportunity to unpack the MLP signal and detect the bits that carry HDCD decoding instructions That's because downmixing, particularly in inexpensive players, employs dither that destroys HDCD code.

What Makes Rotel's HDCD Players Different?

As you've just seen, the HDCD process is a complex one. And, as there are differences in HDCD-encoded discs, there are differences in HDCD-equipped players. Rotel decided to commit to HDCD for two reasons. The first, obviously, was that HDCD brought very valuable sonic benefits to the table. With the growing number of HDCD-encoded discs now available and the process's extraordinary performance with non-encoded discs, we wanted to bring higher fidelity to you.

The second reason was that we knew we could do it better! On the face of it, that's a mighty strong claim. After all, one HDCD-equipped player is much like another, right?

The short answer to that is "No."

First, there are several HDCD-capable decoder/filter ICs available today. Granted, all of them perform the same basic functions. However, not all of them allow the same amount of "tweaking" to match the D/A converters used in players. And not all of them support all the wordlengths and sampling frequencies we'll encounter as HDCD gradually spreads to new formats like DVD-A.

Rotel takes the time to critically listen to all the ways these ICs optimize various D/A converters. We evaluate the effects of different dither types. Listen to different dither levels to critically match the HDCD ICs to the D/A converter we use in every HDCD-equipped player.

Furthermore, we perform Gain Scaling only in the analog domain. So we don't lose digital resolution as some less demanding designs do. And we use only the most overload-resistant components and circuit designs in our analog output stages so you can enjoy the most lifelike, dynamic reproduction of all your music, even those difficult-to-reproduce "Peak Extension On" discs you've just read about.

If all this sounds a bit familiar to you, it should. After all, it's just another example of Rotel's Balanced Design in action. And that benefits you by providing more realism and more musical enjoyment.

And you thought HDCD was just HDCD.

If you want further information on the HDCD process, we suggest visiting the HDCD web site at www.hdcd.com where you'll find all the latest information.

THE ROAD TO BETTER DIGITAL SOUND

Early Digital Demons . . .

The coming of the CD was not a smooth one. In fact, a large number of music lovers began criticizing it as a musically deficient medium almost as soon as it appeared. The more technically astute pointed out the theoretical deficiencies of a digital system using 16-bit word lengths and a sampling frequency of 44.1 kHz.

Many others, too impatient or too ignorant to deal with the theory directly, simply said, "Listen to this garbage!" Early advertising didn't help either as many interpreted the theme of "perfect sound forever" as a cynically commercial and aesthetically insulting threat rather than technology's invitation to a musically better age.

In fact, many criticisms of the sonic integrity of early CDs were well-placed. A large number of discs were audibly more brittle than their analog LP counterparts. Classical music buffs were the most vocal in their complaints. They objected to the sour and metallic rendition of massed strings, the rasping and edgy quality imparted to solo piano and the human voice. Pop and rock fans, for the most part, didn't seem as concerned. (Indeed, with the rise of "heavy metal" music just after the CD's introduction, how would any of them notice?)

Some artists and more than a few recording engineers, however, had serious reservations but these were effectively swept away by digital's advancing tide.

…And Early Exorcisms

Advances in early digital technology were rapid if not revolutionary. In the recording studio, engineers began to re-evaluate and refine common microphone techniques to compensate for the sometimes all-too-revealing and etched quality of early digital processing. Smaller companies developed improved analog filter modules for digital recorders that resulted in a subtle but very real smoothing.

Analog-to-digital converters, responsible for "slicing and dicing" an analog waveform into a digital data stream, underwent substantial refinement. Dithering, or adding a precisely shaped noise spectrum to the digitized audio signal, became more common and helped reduce other harsh artifacts.

On the mastering side, where individual cuts become an album and some production excesses are sometimes countered, design engineers became more aware of the sonic limitations of early processors and began developing refinements to reduce the low-level distortions that plagued the first generation of equipment.

Somewhat surprisingly, advancement on the consumer playback side was much more rapid. Sony, whose first players didn't use "oversampling" digital filters at all, introduced this technology in January, 1985. Philips, an early proponent of digital filtration, soon adopted full 16-bit processing to replace their first generation's 14-bit circuitry.

Engineers, recognizing the importance of a CD player's analog output stage, began developing circuits that better coped with the demanding stepped wavefronts presented by the digital-to-analog converter.

Development and introduction of monolithic ICs with extended 18- and 20-bit capabilities brought on the so-called "bit wars" of the mid to late '80s. High-speed single bit architectures (one marketed under the self-defeating acronym MASH) challenged the established multi-bit designs. The name of the digital-to-analog converter game became "linearity" and numerous graphs (usually produced on Audio Precision's System One analyzer) depicted deviations from theoretical ideals.

Shortly thereafter, several high-end manufacturers "took an axe" to the all-in-one CD player and began offering separate CD transports and stand-alone D/A converters in their quest to better reproduce the information contained on a CD. Digital to analog conversion took on unexpected importance, particularly in the minds of audiophiles.

Today, we are in the midst of a far more profound shift. On the CD-only side, single-box players are again in the ascendancy as separate transports and DACs are increasingly difficult to find. But the major reason for this shift, obviously, is the DVD player. Since all DVD players and drives must also reproduce audio CDs (this *is mandated by DVD standards), and DVD technology is 20 years newer than basic CD circuitry, many of the old design criteria no longer apply.*

Both DVD-A and SACD offer enormous theoretical sonic advantages compared to CD. The key here, as you'll see in the Section on DVD, is that these advantages are mainly theoretical and just don't survive the transition to the real world. Will DVD-A and SACD make the CD obsolete? We think not – particularly not in the next five years or so. Will DVD-A and SACD make HDCD obsolete? Again, we think not. Remember that HDCD scales very well to the 24-bit PCM structure used by DVD-A. (SACD is another matter entirely – HDCD does not work in the DSD arena.)

THE 21ST CENTURY: DATA, DATA EVERYWHERE!

In discussing HDCD, DVD-A, or SACD, we need to remember some basics.

First, a 20-bit word can accurately define 1,048,596 (2^{20}) different sample values. A 24-bit word allows each sample to precisely represent one of over 16 ***million*** possible values. Both data structures represent a substantial improvement over a 16-bit word's 65,536 (2^{16}) possible amplitude levels. That points to substantially less quantization error and spurious noise that might obscure subtle musical details.

Second, a sampling frequency of, for example, 96 kHz means that the highest audio frequency a digital system can accurately process is about 48 kHz — more than twice as high as the 20 kHz CD limit.

Don't get confused here. Remember that the Nyquist Theorem states that the highest audio frequency a digital system can carry is one-half of its sampling frequency. As we've already seen, a CD with a sampling frequency of 44.1 kHz can carry an audio signal up to 22.05 kHz. In reality, most digital systems have a "fudge factor" so the CD's practical upper limit is 20 kHz.

But we can't hear this high anyway. So who cares, right? And if we can't hear to the upper end of a CD's range, why should we even think about the advantages of DVD-A or SACD? (Or about the smaller improvement produced by HDCD processing?)

Surprisingly, the answer to that isn't entirely clear. Although it's true that most of us can't hear steady tones at anywhere near 20 kHz, most or us do respond to instantaneous energy bursts far in excess of that frequency.

No, we don't respond to them as we do to more sustained tones:

Peaks disturb the small hair follicles in the ear canal while steady tones actually deflect the tympanic membrane or eardrum. However, these instantaneous high frequency peaks contain information we subconsciously process as normal components of music. A digital system operating, as CDs do, with an upper limit of 20 kHz thus deprives us of information essential to our perception of music and begins to sound suspiciously unnatural.

HDCD-encoded CDs can convey this high-frequency energy more efficiently. DVD-A and SACD simply do away with this limiting factor.
And the digital beat goes on!

CD-R AND CD-RW

These two "advanced CDs" are very closely related to the conventional CD in that they can carry music in the same form as a conventional CD. But there are two significant differences.

First, CD-R (Compact Disc – Recordable) and CD-RW (Compact Disc – Rewritable) are *recordable* formats, the Digital Age's equivalent to the cassette (or to computer floppies).

Second, although both CD-R and CD-RW *can* record music in digital form, they're both capable of recording many other things, too: Computer word processor files, slide presentations, graphic images, databases, or combinations of different items.

Although you'll find a lot of information about CD-R and CD-RW in these pages, we won't spend a lot of time on non-music applications. There's plenty of stuff about using CD-R and CD-RW for computer applications available on the Internet and we suggest you look there for sources. (Just type "CD-R" or "CD-RW" into any search engine and follow the trail! It's easy.)

The Essential Difference

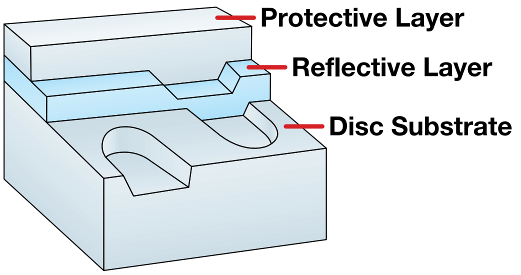
From our perspective, the most important difference between CD-R and CD-RW is that CD-R allows you one chance to "burn" (or "record") the information you want onto a disc. Once you've started the process, you can't change anything you've recorded. You can often (but not always!) *add* information (if the CD-R isn't full, that is) but you can't change or delete anything. So, a mistake puts you in the frustrating position of having to start over with a fresh blank. As we said – frustrating. The advantage here is, obviously, security. This indelible and tamper-proof "write once only" characteristic makes the CD-R ideal for storing and distributing data that you don't want altered in any way.

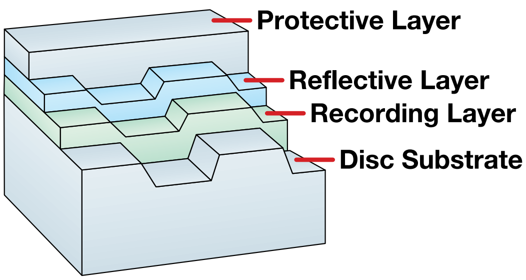
CD-RWs, on the other hand, allow you to go back to the same disc many, many times to add, modify, delete, or otherwise change the information you've previously recorded. Obviously, this is a more flexible format but there is a price – it costs more! And CD-RWs don't provide the data security CD-Rs do.

Introduction to CD-Rs and CD-RWs

In this multimedia age, the CD-R and CD-RW formats has emerged as an important answer to the challenge of efficient data transfer. They're now so popular as to be virtually ubiquitous. CD-Rs and CD-RWs play back on a conventional "music only" CD player as well as CD-ROM and CD-R or -RW drives are now found in almost every computer.

Early DVD drives, however, may not play CD-Rs or CD-RWs. That's because some early DVD laser optical assemblies can't read the reflections from the CD-R's dye layer or the CD-RWs "phase-change" layer. Don't lose any sleep over this. Just be aware that there might be a problem. And be aware that almost all recent DVD players will have no trouble with CD-R or CD-RW discs.





Compared to regular CDs, CD-R technology adds a recording layer composed of organic dyes that respond to heat from a powerful laser.

CD-R Advantages

The CD-R has many advantages for storing and transferring large amounts of data safely and securely.

- **Capacity**: The CD-R has approximately the same storage capacity as a conventional CD — about 650 megabytes. That's the equivalent of more than 450 1.44 meg floppy disks and translates into about 74 minutes of music in what's called "Mode 1" operation.
- **Compatibility**: CD-Rs can be read by both conventional CD players (providing the recorded data is music, that is!), most DVD players, and by computer-based CD-ROM and DVD-ROM drives.
- **Cost**: CD-R blanks are incredible affordable (i.e., cheap!) today. Although consumer disc blanks are more costly than blanks intended for professional applications, they still provide an exceptionally cost-effective way to store data.
- **Security**: Once recorded, data can't be physically erased or changed without destroying the disc. This makes CD-R ideal for archival purposes.
- **Reliability**: During recording and playback, the optical head never touches the CD-R's surface. This prevents damage to the disk and the information it contains. And CD-R blanks are immune to magnetic fields. They don't deteriorate the same way tape media does.
- **Random Access**: Just like a CD, a CD-R disc allows high-speed random access to any band (or file).
- **Distribution Questions**: Recorded CD-Rs are as easy to handle and transport as conventional CDs or DVDs. Their light weight, relative imperviousness to physical damage, the effects of magnetic fields, or abrupt changes in temperature mean that they are ideal for mailing from one place to another. In addition, their high storage density can save substantial time compared to file downloads over the Internet. (**Note**: we strongly caution against exposing a CD-R to high heat and direct sunlight. This combination can affect the dyes used in some CD-Rs and either corrupt previously recorded data or make an unrecorded disc unusable.)

CD-RW Advantages

- Very similar to the CD-R's advantages except that CD-RW adds multiple-record capability. This is a major advantage for most people.

The Consumer vs. Professional Question

Many people are confused about the differences between so-called "professional" CD recorders and discs and their "consumer" cousins.

Briefly, "professional" discs (those able to record 74 minutes or more of CD-quality audio) are specially coded and useable only in machines designed for professional use by recording engineers, etc. There is an added cost in using "consumer" discs as "pro" discs don't carry a "pass along" royalty that goes to the recording industry to compensate them for income supposedly lost because you didn't buy another CD at your local record emporium. However, you can't use a "pro" disc in a consumer machine.

Another advantage enjoyed by "pro" machines is that they don't adhere to something called SCMS (Serial Copy Management System) or "scums." This system, another outgrowth of the record industry's concern for lost revenues, is designed to make it impossible for a consumer machine to make a digital copy of a digital copy.

SCMS works by coding material as protected or unprotected and noting if the source is original or not. The coding uses a single bit that is either always on, always off, or alternating between those two states every five frames. The coding identifies the following:

- Unprotected material, no generation coding: Copy allowed. (The CD-R is also marked "unprotected" so you can make a copy from *it*.)
- Protected material on an "original" disc: Copy allowed. (The CD-R made from this material will be identified as a duplicate and no copies *from this duplicate* are permitted.)
- Protected material on an already-copied (duplicate) disc: No copy allowed. Sorry, you lose!

How The CD-R Works

Recording

A CD-R recorder's laser is the same wavelength (780nm) as that used in a conventional CD player. However, it's about 10 times as powerful, at least when recording. (The power level is scaled back during playback. If it wasn't, you'd be "recording" – or trying to – all over again!)

When recording, the laser's beam is focused on the molded "guide groove" on the substrate side of the disc. (The guide groove helps keep the laser focused properly as it tracks a pre-set spiral that begins at the disc's center and works out.) As the organic dye in the recording layer absorbs the beam's energy, it becomes hot. The temperature rise causes the plastic substrate to deform slightly and the dye to decompose, resulting in a change in reflectivity (the material's ability to reflect light) compared to the unheated portions of the disc. This is the recorded "mark" read by a lower powered laser when the disc is played back.

Playback

Playback, in a conventional CD player (if the data is music), a CD-ROM player, or in the CD-R deck that originally recorded the material in the first place, takes place when the optical circuitry senses changes in the intensity of the reflected laser beam. The reflection is diffused and therefore weaker when the beam hits a recorded "mark" (the equivalent of a "pit" in a conventional CD or CD-ROM) and stronger when the beam reflects off an unrecorded "land."

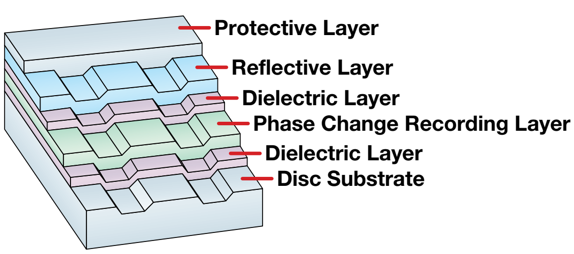
The transient state (when the beam moves from reflective land to diffused "mark" or, conversely, from mark to land) denotes a digital "1," while steady state conditions (whether a strong reflection from a land or a weak reflection from a recorded mark) is designated an "0." Notice that it isn't the mark or the land itself that determines the "1s" and "0s" but the transition between these surfaces that is important!

How CD-RW Works

Recording

You need the following for successful CD-RW recording:

- A hardware platform (either computer or stand-alone CD-RW component).
- Recording software (many choices for computers but usually self-contained in stand-alone components) to convert data to the CD format and to control the CD-RW drive as data is written to the disc.
- A CD-RW disc that meets the standards contained in the Orange Book Part III. (You can use a CD-R blank in a CD-RW recorder *but only once*!)
- A source to be recorded.



The CD-RW is more complex than a CD-R in that it has a recording layer made of phase change material sandwiched between two dielectric – or insulating – layers.

Unlike CD-Rs which use organic dyes that respond to a laser's heat, CD-RW discs use "phase change" materials in the recording layer of the disc. Unlike dyes which undergo a permanent change in reflectivity when hit by a recording laser, the CD-RW's "phase change" chemistry allows the recording layer to cycle between "recorded state" (amorphous stage) and "erased state" (crystalline stage) over 1,000 times before the material does not respond to the recording laser's heat. (Some manufacturers, in fact, claim useful disc life up to 100,000 cycles!) The penalty for this flexibility is that CD-RW discs do not have the comparatively high reflectance level enjoyed by conventional CDs, CD-ROMs, or even CD-R discs.

This gives rise to a minor problem you should be aware of.

The original CD standard (the foundation for CD-ROM and CD-R standards, by the way) did not anticipate "phase change" recording technology or the lower reflectance that the technology implies. This is why some CD and DVD players and CD-ROM drives cannot read CD-RW media. To address this problem, the

Optical Storage Technology Association (OSTA), an optical disc industry association in the United States, has endorsed "MultiRead," a set of standards that ensures compatibility.

The CD-RW's multiple recording capability makes it much more suitable for editing material before it is made into a conventional CD or CD-ROM.

Playback

Playback occurs when the laser beam, now in low-powered "play" mode, follows the molded groove. The reflection stronger as the bean encounters unrecorded or erased sections of the disc and weaker when it "sees" recorded data. As with all CDs, CD-ROMs, and CD-Rs, a change from strong to weak or weak to strong reflection indicates a digital "1" while a constant state indicates a digital "0."

General Notes for Both CD-R and CD-RW

Recording Speed:

CD-R recorders (whether stand-alone components or computer-based transports) can operate at many multiples of "real time." When CD-Rs first became available, high speed "burns" of musical material usually resulted in many errors, some of them quite audible. Recent advances in both recorder and media technology, however, mean that high-speed dubs are, for all purposes, as clean as lower speed burns. You may have to experiment a bit to determine the best trade-offs between speed and error-free operation but your options are all at much higher speeds than they used to be.

Recording Formats:

CD-R supports many recording formats, some with almost cryptic names like RockRidge, Joliet, ISO-9600, etc. These are the most important for our purposes:

- **Audio CD**: This is the format based on the standards defined in the Red Book, established when the Compact Disc came into existence for audio use. The Audio CD format creates discs that will play back on conventional CD players and, naturally enough, is the format you'll be using most often for music recording. If defines recording of a maximum of 74 minutes of stereo, 16-bit, 44.1KHz sampled audio data. .
- ***CD Extra*** (also called Enhanced CD): This relatively new format (codified in 1996) is based on the Red Book (audio CD) standard rather than the Yellow Book CD-ROM standard. It eliminates the problem of earlier Mixed Mode format discs by recording the audio data prior to file data. This allows both types of data to be saved for future use. (Mixed Mode, which predated CD Extra, forced audio data onto the disc after the file data. This caused many problems, particularly very high noise levels, when a Mixed Mode disc was played on a conventional CD player.) A CD Extra disc, then, functions just as an audio CD when played on a regular CD player but allows a CD-ROM drive to retrieve additional information (graphics, notes, etc.).

With both Audio CD and CD Extra, the first step to recording an audio disc involves preparing a sound file that meets CD standards. The WAV format is the standard for the PC world while the AIFF format serves the same purpose for the Macintosh world.

Recording Software:

Unless you're using a component style CD burner (one with two CD trays, one for reading the "source" disc and a separate "burner"), you'll need a good computer and special software to record (or "write") data to a CD-R or CD-RW disc. Here are some common terms you'll need to understand as you look at the available choices.

The "Disc-At-Once" option (a single recording session that includes Lead-In information, data, and then Lead-Out information) uses the entire disc regardless of how much data is actually recorded. This means you can't use any leftover disc capacity. The advantage is that it enables creation of discs that can be played back on almost any CD player or CD-ROM drive because there are no links inserted between the data tracks. This method can also be used to create a premastering disc to be used for mass duplication. (Be careful here if this feature appeals to you. A few early CD-R drives do not support this capability.)

The "Track-At-Once" option, supported by most CD-R and CD-RW drives, allows many separate recording sessions. Remember that a disc will hold a maximum of 30 separate sessions even when no actual data is recorded in each session.

"Packet Write," supported by most CD-R and CD-RW drives, differs from "Disc-At-Once" and "Track-At-Once" approaches in that data is recorded in small fixed sections (usually 64 kilobytes each) called packets. This makes CD-R and CD-RW drives better suited for use as data recorders as opposed to music-only recorders. There are several "Packet Write" formats but they are not important here as they're generally not used for music recording.

Image Files and "On-The-Fly":

You'll see occasional references to "Image Files" and "On-the-Fly" while reading about CD-R and CD-RW recording. They simply mean two very different approaches to creating a file or files you'll need when you custom-record a disc. They use a computer hard drive very differently. Here's a brief overview.

- Image Files: Older computers with slow hard drives and slow processors can cause significant problems when recording. Usually, these problems result in something called a "buffer under-run." This happens when the hard drive or microprocessor can't keep the pre-record buffer filled as the data streams out to the CD-R or CD-RW recorder. This can result in a disc with momentary "blank spots" in the middle of musical information. A temporary "Image File," created prior to recording, often prevents these problems because it is structured differently than the original data file. However, the Image File is as large as the data files so you'll need extra hard disc space to accommodate it (as much as 650 megabytes for a data disc or 740+ megs for an audio CD). Most recording software today has no trouble creating and saving Image Files. Today's computers are usually fast enough in all important parameters so that you needn't be concerned about creating image files.

- On-The-Fly: As you might imagine, "On-The-Fly" is the opposite approach. Rather than creating an Image File, On-The-Fly reads data directly from its original location to send it to the recorder. This requires only a small amount of free hard drive space but is subject to transfer errors if either the hard drive or microprocessor are not "fast" enough to keep up with the bit rate required by the recorder. Again, most computers today support on-the-fly recording with no problems.

You might think you could ignore this distinction altogether if you're just going to record music from commercial CDs, CD-Rs or -RWs onto another CD-R or CD-RW. After all, the material is already formatted and you shouldn't have to use your hard drive at all to make the recording, right? Unfortunately, this isn't always true.

The answer really lies in the software you use to record and the hardware platform you have access to. You may find it advisable to extract the audio information into a WAV or AIFF file first. We strongly advise you to read the software manual carefully and to make sure your hard drive, processor, interface, and CD recorder are all fast enough to allow direct recording without encountering disc-wrecking errors during a session.

Recording Tips

1) You'll need about 10 megabytes of hard disk space for every minute of CD-quality stereo audio you want to record.
2) Most CD-R and CD-RW software programs include a "level meter" display to help set recording levels for analog sources. Watch this display carefully. If the meter goes above 0 dB, you'll be *very* unhappy with the resulting recording.
3) When making WAV files, make sure you use the CD standard setting of 16-bit/ 44.1 kHz.
4) Recording from components connected to your home theater or high-fidelity system is easy. Just connect the "tape out" jacks to your sound card. (You'll need a "Y" adapter – two RCA females to one stereo mini-jack male that will go into your sound card.)
5) The exception here is recording directly from a CD player's digital output to CD-R or CD-RW. The advantage here is that you get an exact digital "clone" of the original recording and you save hard disc space since you don't have to store or reformat any data.
6) If you're recording old LPs, start by cleaning them thoroughly. Keep your

turntable away from the CD-R or CD-RW drive. Vibration from the drive can affect the turntable. If your software has filtering capabilities (and many programs do), use them *carefully* to remove unwanted noise like clicks and pops, etc.

7) In general, we don't recommend live recording directly to a CD-R or CD-RW disc. This is very tough to do well and requires skill, patience, and more than a little luck. Even experienced recording engineers shy away from this practice. An easy substitute is to record to DAT (Digital Audio Tape) and use the tape as a master to prepare the WAV file sequence on your hard drive before recording to CD-R or CD-RW.
8) Remember to "finalize" your recorded CD after you're done. That process writes the TOC (Table of Contents) so a conventional CD player can find different tracks.

Disc Handling Precautions

Note*: If anything, CD-RW discs are a bit more sensitive than CD-Rs. So take even more care in handling them. Remember that the CD-RW's protective layer is only 5 microns (0.005mm) thick so it is very important to* *take care not to scratch the label side* ***or the recording (green) side of the disc.***

That being said, let's look at the CD-R specifically.

CD-Rs are very similar to conventional CDs (audio discs) and CD-ROMs. In fact, the only substantial difference is in the "record layer" of the disc.

Let's look at a conventional disc first. On one side of a 1.2mm thick transparent plastic substrate, you'll find recorded data in the form of "pits" molded into the substrate during the pressing process. There's a reflective layer on top of that (usually aluminum but occasionally gold) and then a protective layer on top of which you'll usually find the disc's label.

Incidentally, you can easily tell a CD-R from a CD or a CD-ROM by the color of the disc's "clear" (or recording) side: A conventional disc is silver or gold while a CD-R has a greenish/yellowish/bluish cast to it depending on exactly what dyes are used in making the recording layer.

The total thickness of the protective layer and the reflective layer is only about 5/1000mm (5 microns) so it's very important to avoid scratching the label side on any disc, whether CD, CD-ROM, CD-R, or CD-RW. In fact, a damaged label side usually means you can kiss a disc goodbye. A damaged substrate side, however, can usually be repaired with careful polishing.

As we've already said, the major difference between two discs (CD/CD-ROM and CD-R/CD-RW) lies in their recording layers. CDs have their recording pits embossed on the substrate during the molding process. The CD-R has recording marks laser-written onto a thin uniform dye layer by the CD-R drive. We've already outlined the essential technology in CD-RW discs.

Be even more careful with unrecorded CD-R blanks than with recorded CD-Rs, CD-RWs, CDs, or CD-ROMs. Because the dyes are formulated to respond to heat, they may respond to direct sunlight and "self record." If that happens, the disc will be useless even though there's no real data on the disc!

Long-Term Storage:

If you're going to use CD-Rs for archival (long-term storage) purposes, the ideal conditions are a temperature range from 60 to 80 degrees Fahrenheit with humidity levels between 40 and 60%.

Compatibility

If you have a **conventional CD** (audio only) **player**, you can read:

1) CD audio discs
2) CD-R and CD-RW discs (as long as they contain 16/44.1 PCM music information)

If you have a **CD-ROM drive**, you can read:

1) CDs
2) CD-ROMs
3) CD-Rs (either music or data)
4) CD-RWs (either music or data)

Note: You'll need MultiRead capability for CD-RW discs

If you have a **CD-R drive**, you can read:

1) CDs
2) CD-ROMs
3) CD-Rs (either music or data)
4) CD-RWs (either music or data)

If you have a **DVD drive**, you can read:

1) CDs
2) DVD-Vs (movies)

In addition, you ***may*** be able to read:

1) **Some** CD-Rs (early generation DVD decks and drives cannot read CD-R discs due to incompatibility between the laser's wavelength and characteristics of the dye used in the discs)

2) Some CD-RWs (for the same reason)

If you have a **DVD-ROM** drive, you can read:

1) CDs
2) CD-ROMs
3) DVD-ROMs

In addition, you ***may*** be able to read:

1) CD-Rs (See above.)
2) CD-RWs (See above.)

Note: DVD-R, DVD-RW, and DVD+RW drives will soon be here and there will be a confusing transition period between various incompatible DVD formats. Remember to check the manufacturer's information carefully before making any recommendations!

M P 3

Introduction

If there's one term guaranteed to raise the eyebrows and tempers of people in the music industry, MP3 is it.

This very powerful compression codec (encoder-decoder) was developed by Germany's Fraunhofer Institute in cooperation with the University of Erlangen in 1987.

Like many lossy compression systems, MP3 was a direct response to the fact that *one second* of CD-quality digital audio requires approximately 1.4 *mega*bits of data. Although CD players have no trouble whatsoever with this high data rate, transferring this data, particularly over the Internet, is an extraordinarily time-consuming operation.

For those who wish to know, MP3 is an abbreviation of the algorithm's full descriptive name – ISO-MPEG Audio Layer 3 where "ISO" stands for "International Standards Organization" and "MPEG" for "Motion Picture Expert Group."

MP3, incidentally, is the most powerful of the codecs developed by the team. MPEG 1 and MPEG 2 (you could call them MP1 and MP2 if you wanted to but these designations are little used) are less efficient in lowering bit rates. On the other hand, they compress the original signal less. Here's a table that will make this a bit clearer.

MPEG Audio	Bit Rate	Compression Ratio
Layer 1	384 kbps	4:1
Layer 2	192 – 256 kbps	8:1 – 6:1
Layer 3 (MP3)	112 – 128 kbps	14:1 – 10:1 (variable – program dependent)

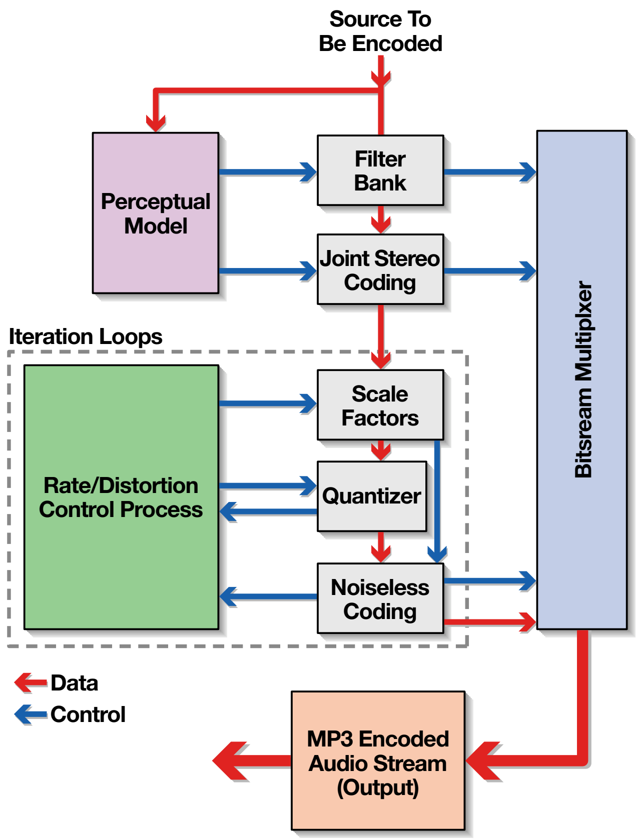
Fraunhofer claims that the MP3 codec can produce "CD quality" sound but, as you'll see in the following table, MP3 falls somewhat short of that mark.

Subjective Sound Quality	Necessary Bandwidth	Mode	Bit Rate	Compression Ratio
Telephone	2.5 kHz	Mono	8 kbps	96:1
AM radio	7.5 kHz	Mono	32 kbps	24:1
FM radio	11 kHz	Stereo	56 – 64 kbps	26:1 – 24:1
CD	> 15 kHz	Stereo	112 – 128 kbps	14:1 – 10:1 (variable – program dependent)

- Bit rates from 56 to 128 kbps (with an emphasis on 128 kbps) are the most popular today. As you can see, the higher the bit rate, the better the sound quality. However, you can find MP3 files encoded at anywhere from 32 to 360 kbps.
- Each frame in an MP3 files begins with something called *header information* – 32 bits of data identifying many attributes of the encoded music – that insures proper decoding.

How MP3 Works

Technically, MP3 encoding is quite complex. Here's a block diagram of an encoder:



\We won't attempt to explain this fully. If we did, you'd soon be awash in terms like "hybrid filter bank," "polyphase filter bank," "Modified Discrete Cosine Transform," etc.

In more understandable terms, the MP3 encoding algorithm uses the following techniques to compress an audio signal. As you'll see, the first two steps depend on perceptual coding that tells us what is audible and what is not. The third is perhaps best explained as a hardware-based "turbo" for demanding musical passages. The fourth step is another implementation of psychoacoustic research while the fifth is simply a more efficient way of "packing" the data for transmission.

- The **Minimal Audition Threshold** tells us what sounds we hear and what sounds we don't and results from a great deal of research into how we perceive what sounds impact our ear drums. Because we don't hear all frequencies at equal intensities, it isn't necessary to encode *all* the audio information. So some of the data is "thrown away" before encoding.

- The **Masking Effect** tells us when a soft tone is masked by a louder sound close to it in frequency. If the soft tone won't be heard anyway, we don't need it. So that data, too, is "thrown away" to improve coding efficiency.

- A **Bit Reservoir** allows efficient encoding of short, intense musical passages that might otherwise exceed the algorithm's ability to compress them without severe distortion. The bit reservoir is contained in a buffer and is available on a short-term basis. The buffered reservoir results from previous encoding of non-demanding musical material.

- **Joint Stereo** encoding occurs when the MP3 algorithm detects essentially the same material in both channels. When this happens, the common data is encoded only once rather than twice (once for each channel) for enhanced efficiency.

- As a final step, MP3 encoders use a **Huffman Code** to effectively shorten the data produced by previous encoding stages. The Huffman Code used in MP3 makes the process 20% more efficient that the perceptual codes alone.

MP3 Today

You'll find MP3 files all over the Internet today. You'll also find MP3 decoding capability built into an increasing number of CD and DVD players today. And, of course, the number of MP3 and MP3PRO (a more advanced version of MP3) software packages for computer use seems to be growing on an almost hourly basis.

The lower bit rate produced by MP3 encoding has two main purposes:

- **You can transmit a music file over the Internet in less than 1/10th the time it would take you to transmit an uncompressed file**. Of course, the price is lower sound quality, particularly at the lower end of the current 64 – 160 kbps data rates you'll now find MP3 files encoded at. In our experience, a 160 kbps file may well be indistinguishable from a CD. Of course this depends on the music itself, the quality of the playback system, and the listener's aural acuity. If you hear someone claiming that their 64 kbps file sounds "just like a CD," you might ask when they last had their hearing checked.

- **You can hold far more musical material on an MP3-encoded CD than you can on a conventional CD.** Combining a CD-R or –RW burner and an MP3 encoder allows a user to place up to 10 hours of custom-programmed music on a blank disc for playback in any of the rapidly-proliferating MP3-capable players now available. Of course, the smaller MP3 file sizes also make it possible to carry a lot of music in solid state memory and there are many portable MP3-capable devices that answer the needs of music-hungry people who don't want to deal with the occasional glitches any disc-based player exhibits when used outdoors, for cycling or jogging, etc.

PREVIEW

We'll look at home theater's origins, see why early experiments in movie sound are so important to us today, try to "crystal ball" the future and identify the concepts you'll need to know regardless of which way this rapidly changing technology will turn next.

INTRODUCTION

Surprisingly, "Home Theater" isn't a new concept. For decades, Hollywood moguls maintained private screening rooms in their homes to show their latest creative efforts. Of course, early "home theater" was strictly an insider's affectation as only those at the top of the film industry's hierarchy had access to prints they could show at home. In addition to the difficulty in getting early software, the technology needed to show it was also a bit daunting. Film projectors were large, complex and cantankerous contraptions. They were noisy and hot – hardly the stuff to inspire confidence or interior decorators! In fact, a separate "projection room" was necessary to both hide the equipment and to isolate the audience from the projector's audible distractions.

Video changed all that.

In 1975, the first consumer VCR (Video Cassette Recorder) appeared. Unimposingly small and relatively quiet, the VCR was nonetheless revolutionary. It turned our televisions into practical time machines: We could watch The Tonight Show at 10 AM, the "Friday Night Fights" the following Wednesday, or the morning news in mid-afternoon. Video cameras, the precursors of today's camcorders, turned our televisions into moving family albums. The arrival of feature length movies on videotape in the mid '70s completed the transformation: The TV became our window on whatever world we wanted to look at rather than a projector for whatever the network programmers wanted to send our way. And the revolution had just begun.

Home theater systems today are the amalgam of many technological influences: the art of high- fidelity music reproduction, advances in film sound and its "consumerization," video hardware (direct view and projection TVs, VCRs, DVDs, videodisc players, satellite receivers, etc.), and the huge variety and easy availability of audio/video software.

HOME THEATER HISTORICAL HIGHLIGHTS

This list of significant developments we take for granted in today's home theater world isn't complete by any means. But it will give you a quick overview of some of the important things.

1909 First experimentation with film sound

1915 First U.S. patent (# 1,124,580) on multi-channel film sound granted to Edward Amet on January 12th.

1927 The ***Jazz Singer***, starring Al Jolson, popularizes the "talkies."

1933 UK Patent # 394,325 granted to A. D. Blumlein on June 14th covering a sound system's vertical imaging requirements for large screen formats.

1941 ***Fantasia***, first publicly shown film with multi-channel soundtrack optically encoded on separate 35 mm film reels.

Harvey Fletcher's influential paper "The Stereophonic Sound-Film System" published in the October issue of the SMPTE Journal.

1958 First commercial stereo LPs.

1961 First commercial stereo FM broadcasts.

1971 ***A Clockwork Orange***, the first film with a soundtrack produced with Dolby "A" noise reduction, opens.

1971-4 The great "Quadraphonic Uprising," a short-lived though exciting battle between technical prowess and musical sensibilities. Fortunately, musical sensibilities aided by a lack of standards and resulting consumer confusion carried the day. Four-channel sound laid the groundwork for the eventual rise of surround sound for video.

1974 ***Callan*** shown at Cannes Film Festival: film is first with Dolby "A" encoding of mono optical soundtrack.

Dolby Stereo introduced at SMPTE Convention.

1975 ***Tommy***, first film with Dolby Stereo soundtrack, debuts in London.

First consumer VCR offered for sale in the U.S.

1976 First film with surround effects, ***A Star is Born***, opens.

1977 ***Star Wars*** debuts in May, ***Close Encounters of the Third Kind*** appears later in year. Dolby Stereo firmly established.

1978 First stereo video cassette.

1979 First VCR with Dolby "B" Noise Reduction shown by JVC.

Apocalypse Now, first film with stereo surround information.

1980 12" Laserdisc format makes first appearance.

Tom Holman joins Lucasfilm to form THX division

1981 Michael Nesmith's ***Elephant Parts*** released: first prerecorded VHS video cassette with Dolby "B" encoded stereo soundtracks.

MTV (Music Television) begins nationwide cable broadcast with Dolby "B"-encoded stereo soundtrack.

1982 First consumer Dolby Surround decoder released.

1984 Dolby Soundlink, first incarnation of "AC-" series of adaptive delta style data compression formats, introduced.

1985 VH-1 begins satellite broadcast using Dolby AC-1 data compression.

First video cassette and videodisc titles using Dolby Surround encoded soundtracks introduced.

1987 First Dolby Pro Logic decoder shown at WCES

Super Bowl XXI broadcast in Dolby Surround

1988 Original THX specifications defined

1991 First demonstration of Dolby Stereo Digital for film industry.

Skywalker Sound begins using Dolby AC-2 coding to link production facilities in Marin County and Los Angeles.

First consumer THX products announced

1992 ***Batman Returns***, first film in Dolby Stereo Digital (AC-3), released.

1993 ***Jurassic Park*** opens – first movie with DTS soundtrack

"Pictures at an Exhibition," first rock recording mixed specifically for Dolby Surround, appears on Emerson, Lake and Palmer's CD ***The Return of the Manticore***.

1994 First appearance of laserdisc with 5.1 channel digital surround soundtrack.

1995 Rotel introduces RDA-985 – first DTS decoder for home use.

1997 Formal introduction of DVD (Digital Versatile Disc)

1999 ***Star Wars: Episode I – The Phantom Menace***, first movie with Surround EX soundtrack, opens in May

First consumer products with THX Surround EX announced (Original THX spec becomes THX Ultra)

THX announces THX Select specifications

DTS announces ES Matrix

2000 Dolby introduces Pro Logic II matrixed surround format; first consumer products shipped.

DTS introduces ES Discrete and Neo:6 matrix surround formats; DTS ES consumer products appear

2001 THX introduces Ultra2 specifications; Dolby announces Dolby Digital EX

2002 Lucasfilm Ltd. spins off THX division as separate company

FILM SOUND

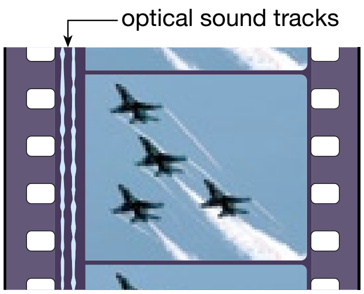
From a purely historical perspective, the first technology to contribute to the home theater explosion is film – specifically, film sound. In fact, every aspect of sound for today's home theater systems began in the movies. That's why it's important to have some understanding of what Hollywood's been doing with sound for the last 80 years or so.

Most people think Alan Crosland's 1927 film, ***The Jazz Singer***, starring Al Jolson, was the first "talkie." Strictly speaking, this isn't true. Although ***The Jazz Singer*** helped focus the public's attention on movie sound, it was really a silent film with a few musical sequences. As for being the first … well, suffice it to say that one-reel movies with crude soundtracks (perhaps better called "audible captions") were first shown about 1909. So much for popular conceptions!

Most of these early efforts relied on large 16" shellac disks played at 78 RPM to deliver the sound. Of course, synchronization between image and audio was somewhat problematic and these "dual media" schemes sometimes ran afoul of rough delivery: One telegram from a movie house owner to the studio read "Film arrived safely, all records broken." Of course, the studio interpreted this to mean that attendance was outstanding while the theater owner was really saying he had no sound source to accompany the movie!

Optical Soundtracks

After some early experiments, the movie industry settled on an optically encoded soundtrack in the early '30s. The optical track was printed directly on the film just as the images were and was therefore far more cost-effective. Because there were no separate records to deal with, there was no chance of loss or breakage – and no synchronization problems between film and disk either!



Printed and developed at the same time as the image frames, optical soundtrack technology brought audio to the movies'.

Film sound in the early '30s evolved very rapidly. The first theater loudspeakers were, as you can imagine, fairly crude devices with poor high-frequency response. Even so, the public demanded sound and the industry scurried to put the new technology into place.

Loudspeakers improved drastically in the first few years of the theater sound explosion (pardon the pun). High-frequency response in particular became far smoother but this fact, coupled with the great demand, created a problem. The sonic discrepancies between older and newer sound systems were so great that the industry had to establish a standard for film sound that actually favored the older, inferior equipment! There simply weren't enough newer speakers (or installation technicians, for that matter) to retrofit the pioneering theaters that had added sound capability first. This was the genesis of the so-called "Academy" frequency response curve.

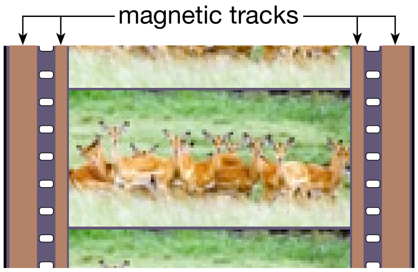
Magnetic Soundtracks

Film sound technology gradually evolved until the '50s when the advent of magnetic soundtracks heralded a significant widescale advance – multi-channel sound.

Like almost everything else in film sound, stereo soundtracks were not an entirely new idea: The first patent covering multi-channel sound for film was actually issued in January, 1915! And ***Fantasia***, Walt Disney's famous 1941 advance in animation techniques, benefited from a multi-channel sound track that was actually optically encoded onto separate reels of 35 mm film synchronized to the image reels. (Does anyone see a parallel here with DTS' movie premiere? If you do, go to the head of the class!)

Magnetic soundtracks were sonically superior to older optical formats, particularly in terms of dynamic range and lower noise floors. However, they were far more costly to implement as strips of fine magnetic particles had to be applied to the film after the image was developed. A final step ran the film/oxide strip laminate past a magnetic recording head to imprint the actual soundtrack on the iron oxide strips. (If this process sounds suspiciously like making a conventional tape recording, it is! Only the size and complexity of the equipment differs from today's cassette recorders as the operating principles are identical.) Another drawback to magnetic technology was fragility. Rough film handling tended to separate the oxide coating from the film base and "dropouts" that caused momentary interruptions became more common as the film ran again and again through the projector. Eventually, this oxide shedding, as it is called, resulted in an unusable print and an expensive replacement.

But magnetic soundtracks provided one inescapable advantage over older optical technology. Film, especially the wide and increasingly popular 70mm film, could hold several magnetic strips. Ah, stereo! Ah, multi-track! Ah, one more competitive advantage over that upstart, television! Hollywood jumped on it.



Multiple magnetic tracks made stereo and surround sound possible even if the technology involved expensive and fragile magnetic tracks.

From a historical perspective, it's important to note that stereo and multi-channel sound didn't evolve in a vacuum. The '50s also saw an incredible explosion of large-screen formats – Cinemiracle, Dimension 150, Cinerama, Cinemascope, and many others – that grew out of the movie industry's concern that the then-infant television networks would siphon off customers. If the televisions of the day were limited to very small screen sizes, the moguls reasoned, why not use the one weapon – a truly large screen – "the enemy" couldn't counter? And so the growth of wide-screen formats was more a pre-emptive strike to enhance the movies' attractiveness than an aesthetic exercise. In the early '50s, fields of view subtended by movie screens extended well beyond the previous 15° standard. Although some formats actually went as far as 180°, most settled around 60°.

The increased field of view brought new challenges for the sound engineers: how to provide some kind of coincidence between the image on the screen – now much more expansive – and the sound. As you might expect, there wasn't any standardized approach here either.

Although some theaters as early as the '30s had three-speaker arrays, primarily to provide smoother coverage over a wide seating area, the '50s really brought multi-channel experimentation to the fore. New formats called for as many as five screen speakers and two effects channels.

These formats used multi-channel capability in various ways. In some cases, multi-channel soundtracks panned the dialog across a five-speaker array in an attempt to synchronize the apparent source of sound with a character as he or she moved from screen right to screen left. Some methods even attempted to provide the illusion of height!

The audible results of some of these well-intentioned ideas were truly bizarre. Vocal origins often jumped in a way that was disturbingly disjointed to those in a theater's prime seats while being barely noticeable to those off-center. Even more astounding were several efforts to add a mixing console in a theater's seating area to control sound quality "on the fly," as it were. Needless to say, most of these methods sank quickly and thankfully into the obscurity they so richly deserved.

Aside from the sonic anomalies, confusion and practical questions from theater owners hard pressed to equip their properties for all the different formats took their toll. The film industry's financial contractions in the '60s ended a lot of this confusion as only a few 35mm formats (including Cinemascope) and one 70mm format survived.

Fortunately for us, this period was much more than just another example of relatively silly "format wars." Extensive experimentation in multi-channel mixing and speaker placement laid the groundwork for home theater as we know it today.

From the Audio Side: Quadraphonic Sound

In the early to mid '70s, the consumer audio industry almost short-circuited itself with a phenomenon called "Quadraphonic Sound." From today's perspective, it is easy to smile at early efforts to convince the public that four was much, much better than two – at least when it came to speakers anyway.

In reality, the period was a curious mix of technology-flexing by developers of matrix systems, manufacturers and the music industry. Indeed, the struggle was tempered by format wars, claims, counter-claims and a real dearth of aesthetic or musical reasons why any consumer would really want to adapt the new technology.

Various four-channel matrixes (QS, SQ, EV, etc.) and one discrete system (called, prophetically enough, CD-4 but restricted to LPs) competed for the public's attention. Confused by, well, confusion, the public largely ignored the whole concept. However, several positive elements emerged from this debacle. First, some far-thinking individuals began to realize that four-channel techniques were perhaps more applicable to video soundtracks than to audio-only sources. Second, the public became more aware that sound reproduction was not necessarily limited to two speakers. Third, the industry realized that no new format would survive in the face of conflicting claims and consumer confusion.

Optical Part II – The Sequel

The allure of optically recorded soundtracks refused to die, however. Economic realities and the somewhat brutal environment inside a movie projector took their toll and magnetic technology lost a lot of supporters.

In 1974, Dolby Laboratories, heretofore involved primarily in promoting its proprietary noise reduction techniques for professional and consumer sound recording, presented the first in a series of developments that were to totally reshape the world of film sound.

Dolby Stereo, first shown at a convention of the Society of Motion Picture and Television Engineers, brought optical technology back to the forefront. Using a technique called stereo variable area (SVA), Dolby Stereo combined the ease and affordability of older optical technologies with the enhanced sound quality and discrete multi-channel capabilities of magnetic recording. It was an instant hit: ***Tommy***, the first film shown in Dolby Stereo, premiered in London early in "75. ***Lisztomania***, the first general feature release, completed production in late "75. ***A Star Is Born***, the first commercial film encoded with surround effects, appeared in the spring of "76.

But it wasn't until May of "77 that ***Star Wars*** finally and indelibly fixed the public's attention on the full potential of this new development. When ***Close Encounters of the Third Kind*** appeared later that same year, it acted as the final catalyst: "Dolby" and "great sound" were cemented together in the movie-going public's mind.

Over the next decade, developments were, well, perhaps less spectacular. At least, it appeared so on the surface. The reality was a bit different. The role of acoustics in film sound became an important area of study and codification. Another was the significant advance in digital signal processing and its promise in bringing even more spectacular sound to the neighborhood Bijou – and to the home.

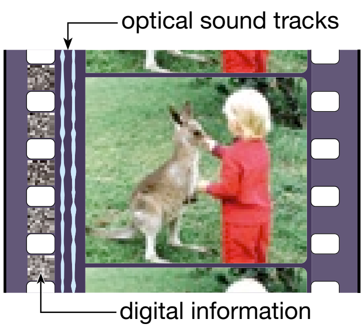
One of the most notable efforts was the gradual rise of Lucasfilm's THX division, an outgrowth of George Lucas' awareness of the importance of film sound in creating a movie's emotional effect. The THX story is an important one for both film sound and home theater.

Another important development was Dolby Laboratory's continuing research into digital sound processing in general and, more specifically, data compression, or the science of using a "quart" container to transport a "gallon" of sound without spilling any.

Although most of Dolby's early work in data compression was aimed at areas other than film sound, the most recent and sophisticated of these compression technologies is one of today's main formats for bringing digital precision to film sound. In addition, Dolby Digital is now the prime mover in carrying digital sound to the consumer market.

The Future of Film Sound: Enter the Digits

Today, the world of film sound is in flux. Everyone agrees that the future is digital. Fewer know which of the three current standards – Dolby Stereo Digital, DTS (from Digital Theater Systems), and Sony's Dynamic Digital Sound (or SDDS) will carry the market or if all three will exist simultaneously.



Film with a Dolby Stereo Digital soundtrack is identical to a standard Dolby Stereo film except for the addition of optically encoded blocks between the sprocket holes. These blocks contain information read by a special head in the projector and processed by decoding electronics in the projection room. Because Dolby Digital films contain all the analog information of conventional releases, they can be played back in any Dolby Stereo theater.

Dolby Stereo Digital uses an optical scanner in the movie projector to read compressed digital information encoded in blocks located between the film's sprocket holes. Dolby Stereo Digital provides Left, Center and Right Front information as well as discrete Left and Right Rear (or Surround) signals and separate LFE (Low Frequency Effects) or subwoofer information. The subwoofer channel carries only frequencies below 200 Hz and thus is limited to a tenth of the normal 20 Hz - 20 kHz audio bandwidth, hence the somewhat confusing 5.1 channel designation.

An often-mentioned component of Dolby Stereo Digital (and, indeed, of all three major digital film sound formats) is data compression. With today's technology, there is simply no way to include six discrete channels of audio information in the physical space left over when the visual image (the "frame") is printed on standard 35mm movie film. In order to provide full surround information, the audio data must be "compressed" to fit.

The **DTS** system is quite different from Dolby's format. DTS is not film-based: a CD-ROM (Compact Disc-Read Only Memory) carries the soundtrack information and plays in a separate transport mechanism. The optical scanner in the film projector needs to read only a SMPTE time code on the film to assure synchronization to within a fifth of a frame. The CD-ROM format provides far more storage area than is available on film. Thus, DTS formatted audio information features a lower level of data compression – about 4:1 compared to Dolby Stereo Digital's variable 10:1 ratio. Specifically, the DTS system produces a digital stream capable of transmitting 1.5 megabits of data per second compared to Dolby Digital's film-standard 320 kilobits per second (kbps). In theory, the denser DTS data stream could result in higher quality sound. However, there are several mitigating circumstances that argue against a significant audible improvement, particularly in theater environments.

Sony's film-based **SDDS** system is much closer to Dolby Digital than to DTS. SDDS places encoded digital sound information on two continuous optical paths located very close to the film's outer edges. The mathematic algorithm (the equation that controls compression and decompression strategy) is a variant on the ATRAC standard first introduced with the MiniDisc and the stated compression ratio is 5:1. SDDS claims a dynamic range in excess of 90 dB which (again, in theory) should prove advantageous.

However, a more interesting characteristic is SDDS's provision for eight discrete channels of information. These two additional "fill" tracks play through separate loudspeakers located behind the movie screen between the normal Left, Center and Right Front speakers. While not a new idea (you'll remember that 70mm film used this technique about three decades previously), it carries some potential for improving localization, particularly in very large theaters. For compatibility with conventionally equipped or smaller theaters, the eight channels can be "folded down" to six or four.

Data Compression: Can We Hear It?

Before we go any further, we need to take a closer look at data compression, which involves complex analysis of an incoming signal to remove so-called "unnecessary" bits so we can better fit data into the space available to contain it. This technology has become audio's "Bad Boy" in the last few years. The high-end audio community in particular sees data compression as a generic attack on the integrity of any signal. The consequent penalty, according to them, is distinctly lowered sound quality. On the other hand, engineers with different sensibilities see data compression as an effective way to bring the potential of digital sound to the broadest audience.

There is no hard and fast answer to the question "Is it good or not?" Data compression's effects on an audio signal depend entirely on the algorithms used to determine which bits can be safely (i.e., non-audibly) "thrown away." The psychoacoustic models used to structure these equations must be accurately researched and there is simply no way for a consumer to fully determine their efficiency.

Careful listening is the only real method open to us and, even then, stringent controls must be used for proper evaluation. One thing is for sure: data compression is here to stay. Today's encoding strategies are much better than those of ten years ago and we see little chance of that progress abating.

Making Sense of the Claims

Briefly stated, compression means altering digital data in some way so that it occupies a space it couldn't "fit" into before it was compressed.

Of course, a thorough technical discussion of compression would be almost hopelessly daunting so we'd like to suggest an easy way to think of it.

If you had to transport a load of bricks in one trip with a pickup truck, there are two general ways you could go about it.

You could just throw the bricks into the back of the truck willy-nilly and make do with whatever number you could cram in. Of course, that's not very efficient. For one thing, there would be no way to guarantee that you'd be able to transport all the bricks you needed for that home improvement project. And there's no guarantee that you wouldn't leave the most important bricks at the brickyard! (Tough, Charlie, once you left 'em, they're gone!)

Maybe the most elegant solution is to carefully pack the pickup so that all the bricks fit. Sure, this approach means more "packing time" but it also means that all the bricks get to their destination in one trip!

Of course, you could solve the problem by getting a bigger truck. But that would cost more and you'd still have to worry about the pickup, wouldn't you?

Once you understand these examples, there isn't much more you need to know – at least conceptually – about data compression.

Have You Lost Anything?

When we're discussing compression, we describe these options a bit differently. But they're essentially the same. For one thing, we note that there are two types of compression – "lossy" and "lossless."

"Lossy" compression makes the load smaller by intentionally leaving some data out.

"Lossless" compression simply packs the data more efficiently so that you can get it into whatever you have to "carry" it.

Most of the compression we use for audio is lossy. Dolby Digital depends on lossy techniques. So does DTS, if to a lesser degree. MP3, the Internet favorite, is very lossy – and sounds it!

Not all lossy compression is the same. Some forms are very good in that they provide enough aural information so that music sounds surprisingly natural. But some lossy compression schemes are not that kind to the original signal.

Most lossy techniques depend on something called "perceptual coding" for their effectiveness. Without getting too bogged down, perceptual coding is based on a psychoacoustic principle called masking. Masking tells us that our ears really don't "hear" a low-intensity sound if there's a louder sound very close to it in frequency. (The louder sound "masks" the softer.)

Lossy compression builds on this principle and says, in effect, "If you can't hear it, you don't need it so let's just throw it away!" The problem here is that once data is tossed into the Giant Electronic Trash Can In The Sky, it's gone – and can't be recovered.

In truth, lossy compression is totally unsuitable for archival use – preserving musical performances for future generations, for example. But lossy compression is a valuable tool for any application short of the absolutely critical and can be used successfully when "good enough" is, in fact, "good enough." And sometimes "good enough" is far more than just adequate.

Lossless compression is obviously different. To begin with, it means exactly what it says – there is no data lost in the compression process. It is the digital equivalent of meticulously placing each brick in the back of the pickup truck so that every cubic inch of space is packed.

Any of you familiar with a computer utility called PKzip or its newer cousin, WINzip, knows how effectively these programs work. They can compress a file by up to 80% and then reconstruct it so that it is bit-for-bit identical with the original! That means that there is no data loss whatsoever and that the file sent through the compress/uncompress cycle is every bit as useful as the first, uncompressed file.

Home Theater: A "Work In Progress"

Home theater is an evolving concept. Today, there are exciting developments on both the hardware and software fronts that will make home theater one of the most cohesive and powerful concepts the consumer electronics industry has ever seen.

The broad brush strokes of home theater's hardware path are already on canvas. Indeed, we're now experiencing:

1) The meteoric rise of DVD (DVD for "digital *versatile* disc," by the way!) as *the* premium program source for those wanting more complete control of what they watch as well as exceptional surround sound capabilities.
2) The continued growth of small-dish satellite receivers as a popular program source
3) Even more use of Dolby Digital (and, to a much lesser extent, DTS) as a discrete surround sound delivery method. Of course, Dolby Digital already enjoys its status as the required format for DVDs, and is also the standard for DTV (digital television) broadcasts.
4) Progress past the current "5.1" surround sound standard. Of course, this will bring with it a lot of confusion (is Dolby Digital EX really "6.1" or just "5.1+"?).
5) The eventual arrival of DTV as a viable source of video programming. .

The Future of Digital Audio . . .

A side issue to the rise of DVD is the obvious presence of DVD-A and SACD. We'll cover both of these exciting formats in the separate section on DVD. In the meantime, we'll just say that both formats offer a sonic potential far in excess of the CD. The winner (and we frankly wonder if there *WILL* be a winner in this race) will set yet another standard for our aural expectations.

Beyond this is the prospect of exactly how we are going to get entertainment software into our homes in the future. Expanded "on-demand" services, both cable and satellite based, are just beginning to supply a wide choice of movies and music on a continuous basis. Fiber optic cable technology will grow significantly in the coming years – as will satellite broadcasts - to provide more program options with improved audio and video quality.

Although several legal battles have temporarily squashed web sites like MP3.com, file sharing on the Internet will continue to evolve as the music and film industries take advantage of these new distribution channels. We expect to see even more effective compression that will eventually allow full-motion video to stream into a viewer's home over conventional phone lines. The rise of broadband Internet access will also contribute to more convenient program availability.

And don't forget electronic games. As foreign as they are to many of us in the audio/video industry, they will also help shape home theater's future. Game programmers have long relied on sound effects as an intrinsic part of their creative arsenal. More recently, game programmers have taken their cues from the movie industry and realized the strong appeal of surrounding game players with sound effects to supplement the video image. Companies like Dolby Laboratories are moving strongly to position themselves in this new market and gamers will, in turn, eventually influence the more traditional markets as well.

FIELD OF VIEW

One of the most important and misunderstood aspects of home theater enjoyment is field of view. More concretely stated, "field of view," measured in degrees, tells us over how wide an angle our vision covers.

For most of us, binocular vision covers about 110° on the horizontal plane. We don't use this entire field of view equally. In fact, when something attracts our attention at the periphery, here defined as anything beyond 40° to our limit of 110°, we usually turn our heads in that direction to focus full attention on whatever triggered our interest.

Early movie screens rarely exceeded a 15° field of view. Even specially constructed "movie" theaters (as opposed to converted "legitimate" theaters) barely reached 25°. Consequently, audiences never had to move their heads to follow the action. In fact, such a narrow field of view meant that members of the audience didn't even have to move their eyes to take in the entire image.

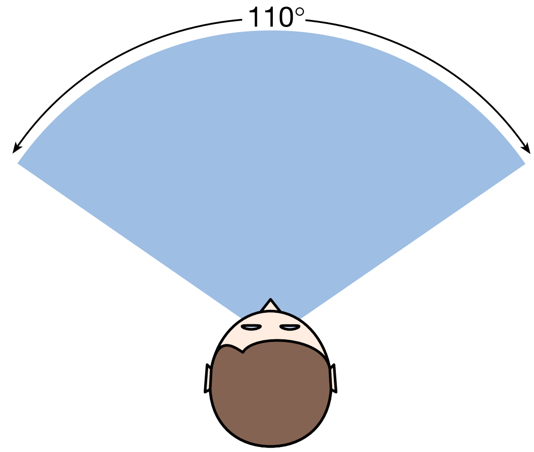
It is interesting to note that modern television rarely subtends an angle greater than 15° either. This is more than an academic note as it directly relates to home theater setup. Of course, there were exceptions to this rule as many early film directors, attracted to the way very large screens "involved" the audience, experimented with different approaches. In 1900, for example, the Paris International Exhibition featured a film shown on a screen 100 meters wide. (For reference, consider the fact that a football field is only slightly smaller!)

Because film stock was comparatively narrow and projector optics were still relatively crude, these early wide-screen efforts required multiple synchronized projectors. In fact, the Paris screening used 10 and the final segment of Abel Grace's monumental 1929 ***Napoleon*** required three.

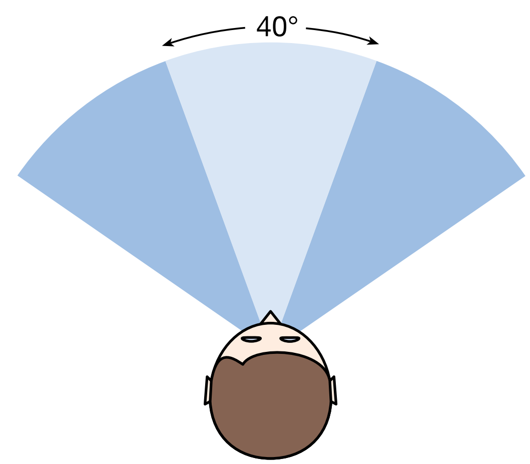
In the late '50s, several very wide screen formats (CinemaScope and Todd AO, for example) provided horizontal fields of view in excess of 60° when measured from a theater's prime seats. This regularly required members of the audience to move their heads to focus on action beginning on the edge of the screen and the frequent head motion proved mildly disconcerting in a theater environment.

Today's 35mm film standard subtends an angle of about 45°. This sometimes requires a minimal amount of head movement and often forces our eyes "off center" to concentrate on some developments. Rather than being a distraction, the minimal head movement emulates the real world and makes movie viewing more natural. The 45° standard also has direct bearing on the way film sound is mixed and is an optimal angle for setting up speakers in a home theater system.

Field of view also plays a major role in determining how far from the screen we sit in a home environment. Until DTV or, even better, HDTV becomes widespread, we're never going to approach that "ideal" 45° viewing angle with conventional video sources. With few exceptions (systems using video line enhancers are among the only ones), NTSC resolution standards are simply too low.



Normal eyes register movement and shapes within the blue area. We say our field of view 'subtends' an angle of 110°.



If movement or a shape in the dark blue area (outside of the 40° field of view angle) attracts our attention, we usually turn our heads towards the point of interest.

PREVIEW

This Section begins "the heavy stuff." But don't worry, it'll be enjoyable anyway. Here are some of the things you'll find:

1) How to discover exactly how exciting home theater really is.
2) How home theater components work together and how to understand the technology in simple and non-confusing ways.
3) How the audio side of home theater is similar in many ways to good high-BN Zfidelity sound reproduction and how it's different.
4) How to plan a home theater system

We've also included some buying and selling tips we've found helpful in convincing yourself – or your customer – to begin enjoying home theater as quickly as possible.

EXCITEMENT HAS ITS BENEFITS!

Properly chosen and correctly installed, today's home theater systems put the excitement of Hollywood's best films in our homes. In fact, if there's any one word that really describes – and sells – home theater, it is "excitement."

The technology is really secondary. Although that sounds like heresy, it is certain truth. We're not saying technology is unimportant. It is. But today's customers buy *benefits* — how technology changes their lives for the better — rather than specific circuits that produce a particular effect.

For that reason, we've written this section from a very practical viewpoint. Yes, you'll find technology here but we've intentionally focused on making it more understandable and less intimidating for consumers and salespeople alike.

Of course, we're prejudiced in favor of good sound. We fully believe that it adds the most important element to any home theater system. From our point of view, two facts stand out:

1) *Regardless of the source, present and future home theater systems will depend on superb audio reproduction for their impact and real emotional appeal.*
2) *This audio information will come from more than just two speakers.*

DEFINING HOME THEATER

Defining home theater today is perhaps a bit more difficult today than it was only a few years ago. Most people accept home theater to mean a home entertainment system with the following:

1) A larger-than-normal television: 27" or larger for a direct view set and over 40" if a projection model.
2) A sound system with digital surround capability and at least 5 speakers (subwoofer optional). The audio components are connected to the video gear

so that audio soundtracks of video sources play through the system's speaker array rather than through TV speakers.

3) Video sources – VCRs, DVD players, cable or satellite STBs (set top boxes) – that allow you to watch what you want when you want. Sound systems can take many forms, from "all-in-one" home-theater-in-a-box systems through carefully chosen and matched component systems usually sold complete with convenient racks to high-performance individual components.

Rotel obviously favors high-performance individual components because:

- They are usually built to a higher quality standard than are all-in-one systems. That means more advanced engineering, higher parts quality, and tighter manufacturing tolerances.
- They result in better flexibility: customers can choose, with the help of competent salespeople, exactly the components that best suit their needs.
- They allow easier and more affordable upgrading as consumers must replace only the component that no longer suits his or her needs. There's no need to replace an entire system.
- They minimize downtime and inconvenience if repairs are ever needed. If one component malfunctions, only that unit needs to be looked at by a service technician. The rest of the system continues to offer enjoyment.

HOME THEATER COMPONENTS

Let's begin with a brief overview of the components of a home theater system.

The Sights . . . TVs

Most of the video information in this section of the Second Edition concerns NTSC-standard sets and broadcasts. (NTSC, incidentally, stands for "National Television System Committee," not "Never The Same Color" as some wits would have it.) We've enjoyed "NTSC video" for decades. However, the combination of interlaced picture technology, 525 scanning lines, and a 4:3 aspect ratio means that NTSC pictures can compete visually with the "film-like" quality the new high- definition digital TV standards can provide.

Digital television (or DTV) promises us better picture quality than NTSC broadcasts but it will come a bit more slowly to most of North America than originally thought.

For now, let's look at televisions a slightly different way – by how they produce a viewable image. Here, you find Direct View, Rear Projectors (usually one-piece designs), two-piece Front Projectors, and Flat Screen TVs.

Direct View Sets

These are conventional color TVs sets that use picture tubes, more accurately referred to as cathode ray tubes – or CRTs. Direct view sets offer the following advantages:

- **Familiarity**: Everybody knows what they are. Everyone feels comfortable with them because there's nothing new or off-putting here.
- **Picture quality**: Direct view sets still have excellent definition, color saturation, etc. Projectors, particularly when used with an image-enhancing device (a line doubler, quadrupler, etc.) can be even better but they're more complex and certainly more expensive.
- **Seating flexibility**: Direct view sets allow you to sit almost anywhere in a room and still see a sharp, color-balanced picture. Projectors, particularly rear projectors, don't offer as much flexibility here as their contrast and color balance deteriorates rapidly as you move off-axis.
- **Will work better under different conditions**: Nothing washes out the picture of a projector set more quickly than bright ambient light in the viewing room. Direct view TVs are far more immune to this interference and can be enjoyed in a wide variety of lighting intensities.

Measured diagonally, direct view picture tubes range anywhere from below 12" to 40". For home theater use, anything smaller than 27" probably isn't an ideal choice simply because the smaller image isn't as emotionally involving as a larger one would be.

Direct view sets are not all created equal. Some exhibit much better color saturation and "black level" rendition than others. Some have greater resolution capabilities although the real world benefits of "700 lines" of resolution are somewhat dubious as only DVD among common video sources goes significantly past 400. (NTSC broadcast standards call for 525 "lines" of vertical resolution. However, some of these lines carry guard band and synch information and don't contribute to picture quality *per se*. In addition, typical distribution problems with over-the-air transmissions or through cable networks effectively reduce performance to even lower equivalent levels.

Rear Projector Sets

Although some fairly esoteric rear projector sets could be described as "two-piece" displays, we're going to use the term "rear projector" in its most accepted way, i.e., one-piece cabinet-style sets.

Rear projection sets have a major advantage over direct view sets: **They usually provide a much larger and therefore more dramatic image than direct view models.** While most rear projection sets average 50", some models have screens in excess of 80" (measured diagonally, of course!) That's double the diagonal measurement of the largest direct view set available today and, as you can easily imagine, presents an irresistibly dramatic focal point for a home theater system.

But the large image is not a "free lunch." Rear projectors have three major disadvantages compared to direct view sets:

- **Reduced contrast and color fidelity** even when viewed on-axis.
- **Further deterioration** when viewed **off-axis.**
- Greater sensitivity to ambient light which results in **picture "washout" when viewed in a bright room.**

Most of these problems result from the fact that the screen in a typical rear projector design must compensate for the comparatively low light levels thrown by the projector elements. Because the available light per square inch is considerably less than produced by a direct view set's cathode ray tube, the picture lacks the brilliance and immediacy we sometimes expect.

Another caution is that the comparatively large projector elements can come out of alignment during shipping. Although "convergence" adjustments often solve this problem, they must be done in a thoroughly meticulous manner by a skilled technician.

Despite these potential cautions, the size of the final image is often the single most important purchase consideration. Here the rear projector has a definite advantage.

Note: New technology, particularly DLP (digital light processing) developed by Texas Instruments, is rapidly gaining headway in the rear projector market. In addition to very impressive image quality, DLP displays are so thin that many people confuse them with the so-called "flat panel" displays you'll read about shortly. In reality, of course, DLP displays are really rear projector sets (even if they take up far less room than conventional rear projectors) and are not as thin as their true "flat panel" LCD and plasma cousins.

Front Projectors

Front projectors consist of a projector unit and a separate screen. The projectors themselves can use CRTs, LCDs, or even the newest DLP (digital light processing) "mirror" technology. A dealer will be able to explain the theoretical and practical difference between these and other types of front projectors.

Front projectors, especially those using CRTs, are considered the king of the video image makers for home theater. Although still performance-limited by available light, they are capable of truly breathtaking performance when used in a dark room.

One of the main reasons front projectors are popular in high-end home theater systems is the ease with which they mate with video processors (so-called "line doublers," "line quadruplers," interpolators, etc.) that overcome many limitations of the NTSC standard. Although not fully up to film's resolution capabilities, a "doubled" or "quadrupled" image shown through a properly set up front projector can be an enthralling experience.

Remember that the **screen** will have a major role in determining overall picture quality. This is a complex topic all to itself and beyond our scope here. However, don't think it's unimportant!

Flat Panel Display Devices

These are *not* the conventional TVs you see advertised as having a "flat" picture tube! Instead, these are the "hang on a wall" screens we've all read about for years now.

Flat panel displays, which use either LCD or "plasma" technology, are very thin – rarely more than 4" to 6" deep. Thus, they're ideal for the décor-conscious home theater aficionado as they are the least intrusive way to get a big image (currently up to 62" measured diagonally) into a well-furnished room.

And they're a reality today. However, as with any comparatively new technology, there are a few things to keep in mind if you're considering one.

- **They're generally relatively expensive** – especially those using "plasma" technology. (Prices are declining rapidly, though, so you may want to do some research before coming to any real conclusions.)
- **They're "fixed pixel" devices** that have to be carefully set up to get the most out of your favorite video sources. You should know a few things about such esoterica as "native resolution" to choose the best flat panel display for your needs.
- Despite rapid improvements in reliability, flat panel displays **still aren't as "bulletproof"** as conventional CRT or projector TVs.

Video Source Components

VCRs

Despite the current hoopla over DVD players (and their almost unbelievable increase in popularity), VCRs are still the single most prevalent source for "home theater-style" viewing – although they might not be for long, thanks to run-away sales of DVD players.

Currently, the number of choices facing the customer for a VCR is narrowing. Indeed, VCRs have become "commodity" items now with little differentiation. If you're buying or selling a VCR for home theater use, "Hi Fi" capability is a must. An advance over older (and now-obsolete) "stereo" VCRs, Hi Fi VCRs record two full-range audio channels "squeezed" between the black and white (luminance) and color (chrominance) portions of the video signal. A "Hi Fi" VCR uses FM (Frequency Modulation) and records audio data on the wide tape area covered by the helical scan video head assembly. The result is excellent sound with superior frequency response, wide dynamic range, and very low noise.

On the video front, however, most VCRs suffer a distinct performance disadvantage in comparison to the best NTSC broadcasts. Typically restricted to about 240 horizontal lines (or "pixels") of resolution, standard VCRs don't even come close to NTSC's 330 lines of effective visible resolution. S-Video circuits (with their separate signal paths for luminance and chrominance) can increase picture resolution substantially, however. The best S-Video units push more than 350 lines of resolution with special high-density tape formulations.

DVD Players

Because the Second Edition of this book contains whole new sections on DVD, we're just going to give a brief overview here.

With the official (and somewhat silly) name of Digital Versatile Disc, this CD-sized format combines both digital audio and full motion digital video for superb playback quality.

DVD standards cover many applications. Movie playback, our prime concern here, is only one. Others include DVD-ROM (DVD – Read Only Memory) and a host of confusing and incompatible "alphabit-soup" (pardon the pun) derivatives for computer and recording applications, a game standard, and DVD-A, a high-resolution "audio-only" standard that takes advantage of a DVD's enormous storage capacity.

DVD is fully digital in nature. (The old 12" laserdisc carried video information in the analog domain.) In order to get enough information on the small disc for 2+ hours of high-quality playback, both audio and video data are compressed. Video compression follows standards set down by the Motion Picture Experts Group, hence the acronym MPEG-2. The mandated multi-channel audio format is Dolby Digital. (Again, you'll find more about all of this in the DVD Section.)

DVD's major advantages include convenience and quality. The public has already accepted CD and a video-specific format of the same size has enormous appeal. Horizontal resolution already exceeds 400 lines or pixels and further advances in video compression will yield even higher line counts. But more significant is the improvement in video signal-to-noise ratio. This allows the small disc to play back more intense colors with less streaking and far fewer momentary dropouts we've become accustomed to in today's video sources.

On the audio side, six-channel digital sound (five full-range channels and one low-frequency-only channel) approach effective 20-bit resolution (compared to the CD's 16-bit standard) even though the various data compression/expansion strategies have some impact on final sound quality.

Laserdisc Players (A Historical Note)

The 12" laserdisc (LD), now relegated to footnote status by DVD and valued only by collectors today, sits at the crossover point between analog and digital technologies.

Even though the LD uses a spiral of pits much like that of a CD to store information, the LD's video data is analog rather than digital. But the LD's picture quality is still formidable with up to 400 lines of horizontal resolution. In fact, the laserdisc carries an even more detailed signal than the best NTSC broadcasts (even if it is not quite their equal in signal-to-noise performance).

The LD's flexibility and sound capabilities were also extraordinary. Towards the end of this format's useful life, digital memory techniques had largely erased the special effects differences between CAV (constant angular velocity) and CLV (constant linear velocity) discs. (Don't ask about these differences – you don't need to know!) Concurrent mechanical advances meant that many later LD players could scan both sides of a disc for uninterrupted playback of a two-hour movie.

Although early LD discs and players employed analog audio only, players and discs manufactured in 1984 and afterwards were capable of both analog and digital audio. These later players could read audio information in two ways, either from standard two-channel analog audio tracks using the same type of frequency modulation as Hi Fi VCRs or from full PCM (Pulse Code Modulation) digital audio tracks that produced CD-quality stereo sound. All LD players manufactured since 1984 automatically default to the digital audio data rather than the lower quality AFM analog audio information.

In addition, a few LD players also included Dolby Digital circuitry that read a full 5.1 channel discrete Dolby Digital soundtrack – provided, of course, that the disc itself carried the Dolby Digital information in the first place. (These Dolby Digital discs sacrificed one of the analog FM tracks in order to make room for the Dolby Digital information.)

Thus, an LD today can contain either:

1) two separate AFM analog tracks and one serial PCM data stream or
2) one AFM track (in mono), the PCM data, and a Dolby Digital-encoded digital bitstream in place of the sacrificed AFM track.

Digital Theater Systems (DTS) had another approach to the LD. Their idea was to sacrifice the serial PCM track and replace it with a high-density six-channel digital signal that, in theory, promises sound at least as good as Dolby Digital. However, this meant that a standard (i.e., non-DTS equipped) LD player would revert to the lower quality AFM tracks for its audio data as the disc would have no PCM data stream available. As you can imagine, this did not sit well with many users and DTS's inability to bring discs to market made their system a practical non-entity.

Satellite Boxes and Cable Converters

With the exception of so-called "premium" channels like HBO and Cinemax,most people overlook cable as a source for home theater programming. That's a mistake. Many consumers are very comfortable with cable. Recent competition from satellite providers has pushed the cable companies into providing far more extensive "on-demand" programming than they had previously offered. In addition, some cablecasters are upgrading their distribution networks with fiber optic links for significantly better picture and sound capabilities which, in some cases, are sorely needed.

The rapid emergence and consumer acceptance of small dish satellite formats like DirectTV demonstrates surprisingly high "comfort levels" with this new format. These digital transmissions combine very good picture resolution with excellent audio quality.

Remember Dolby Pro Logic II decoding when thinking about these sources. Although the satellite industry has moved rapidly towards full Dolby Digital broadcasts, there are still many with Dolby Surround information only. Although Dolby Digital processors will produce an acceptable surround experience with a Dolby Surround source, some involved professionals still prefer Pro Logic (particularly the new "II" version) when dealing with these admittedly older soundtracks.

Another thing to consider is that many satellite sources and a slowly increasing number of cable companies are now supplying high definition video (HDTV) signals to be enjoyed by those with home systems able to take advantage of this new standard. We see increasing use of satellite and cable as a dependable source for truly satisfactory picture (if not programming) quality.

NTSC Video: Getting the Signal from One Unit to Another

In addition to the standard composite video RCA-style connection, many higher quality video components – certainly many DVD players and even some VCRs – use a multi-pin "S-video" connector to transmit video data. And many DVD players now include "component video" connections that make use of 3 RCA jacks.

S-Video connections use multiple conductors to keep the luminance ("Y" or black and white portions of the video signal) and chrominance ("C" or color information) separate. This is in sharp contrast to the more common "composite" video signals where luminance and chrominance are mixed and transmitted through coaxial RCA "patchcords."

S-video connections can avoid one problem common to video signal transmission. If luminance and chrominance components are separated throughout the video signal chain, "downstream" components don't have to separate these elements again. This usually results in a visible improvement in picture quality. However, if the luminance and chrominance components are ever combined into a composite signal, subsequent transmission using S-video connectors results in only barely noticeable improvements.

Component Video connections are now standard on a growing number of DVD players, other source components, and monitors. "Component video" carries separation one step further. Actually a "consumerized" version of professional video's RGB (Red, Green, Blue) standard, component video provides even better color saturation. All else being equal, component video adds a "film-like" quality to video playback that is quite impressive. Of course, to take advantage of component video's potential, your display device (direct view or projector) must have component inputs.

Digital Video: Getting the Signal from One Unit to Another

Here things are much more problematic – or at least much more confusing. At the present time, "content providers" (Hollywood, TV programmers, etc.) are very concerned about piracy. Although there are several methods of getting a digital video signal from one component to another (from a DVD player or cable box, for example, to a display device), there is no general agreement about how this will be done.

The problem is that an "unencrypted" (read unprotected) signal could be shanghaied, duplicated, and then distributed without any financial benefit accruing to the originator of the material.

Although technologies like "Firewire" (also known by the engineering standard IEEE 1394 or the trade name "iLink") are more than capable of transferring even high-definition video signals easily, the protection questions are, at least of this writing, still largely unresolved.

So, if you were to ask us how you could get a digital video signal from one place to another, we have an easy answer for you. You can't. At least not yet.

A composite video cable looks just like an audio cable. However, both the cable itself and the connector are optimized for video.

A single S-Video cable is actually several cables in one. That's because the S-Video format separates brightness and color information.

You'll need three separate signal paths to get one component video signal from your DVD player, for example, to your monitor.

PREVIEW

This Section examines the differences – and the similarities - between home theater and music-only systems.

Are these differences real? If so, where do they come from?

Which ones are truly important? Are there optimal solutions that can make one system equally able to reproduce music and soundtracks?

To answer these questions, we'll look at some of the acoustic realities governing all sound reproduction systems – whether audio-only or audio-for-video. We'll give you some easy-to-follow guidelines as you look for the best – and most flexible – "budget friendly" systems.

INTRODUCTION: DEFINING THE ISSUES

According to some audiophiles, home theater systems and accurate music reproduction are somewhat at odds. A few of the things mentioned to advance this argument are well worth looking at. You'll need to understand them to effectively present home theater concepts to your customers (if you're a salesperson) or to understand why a salesperson is offering particular types of advice (if you're a customer).

Here are the major issues we'll be exploring:

1) The need for three front speakers (as opposed to conventional stereo's two)
2) The "Imaging" question (or how speakers present a convincing sense of "there-ness")
3) The "Ambience" question (or what constitutes believable "surround sound.")

An undercurrent runs through this discussion. It has to do with standards. The fact is that the world of film sound must adhere to several very well-defined and almost universally accepted guidelines. The audio-only world is almost entirely devoid of commonly accepted acoustic parameters for recording and playback. This leads to several major problems. If there's no agreed-upon standard for the "sound" of a recording studio control room, for example, what's the reference for determining what the artist intended in the first place?

One good aspect of this uncertainty is the almost unlimited room for discussion among recording engineers and audiophiles. One bad aspect is that the vehemence of this discussion often precludes any rational pursuit of real understanding.

GRAMMY VS. OSCAR: MAY THE BEST SOUND WIN!

The Oscar for Best Sound, the Holy Grail of the film sound world, is awarded by qualified members of the Academy of Motion Picture Arts and Sciences (AMPAS). People voting in this competition all listen to the nominated soundtracks in similar, if not identical, acoustical environments that conform to a standard called SMPTE

202M. (SMPTE stands for the Society of Motion Picture and Television Engineers.)

The Grammy Awards for Best Engineered Recordings, in contrast, are granted by NARAS (National Academy of Recording Arts and Sciences) after many individual "at home" auditions of the nominated albums. This means there is no standardization of either playback equipment or listening room acoustics.

Don't take this to mean that one group is more "advanced" or more correct than the other. It simply underscores the differences between the audio and film sound communities.

This difference accounted for the frustration felt by some high-end audio manufacturers and hobbyist publications as home theater began to take significant market share in the early '90s.

Home theater technology was largely developed by organizations like Dolby Laboratories, the THX division of Lucasfilm Ltd., and DTS. Some of the smaller and more iconoclastic "high-end" audio manufacturers were slow to adopt these new formats and often railed extensively about home theater's "evils." This isn't to say that their concerns were simply narcissistic. In fact, some of the ideals guiding the home theater community (particularly those dealing with loudspeaker dispersion) hampered the creation of a believable "soundstage," a quality long held in high esteem by knowledgeable audiophiles.

However, market forces prevailed. Today, almost all manufacturers offer multi-channel processors and amplifiers. And the questions of loudspeaker dispersion have been answered either by a relaxation of early home theater standards or by a better understanding of how to meet these often-conflicting requirements.

SMPTE 202M: THE STANDARD FOR MOVIE SOUND

SMPTE 202M, 9 pages of fairly dense technical material, is the film industry's "Bible" on the sonic qualities of all recording and dubbing facilities used to produce movie soundtracks. Last revised in 1991, SMPTE 202M's official title is "B Chain ElectroAcoustical Response — Dubbing Theaters, Review Rooms, and Indoor Theaters."

SMPTE 202M covers many areas but its most important contribution is that it standardizes frequency (or amplitude) response. This means that the tonal character of movie soundtracks will not change radically from dubbing stage to the neighborhood Bijou.

Although the balance of 202M's exact parameters are beyond the scope of this Guide, it is important to note that the music-only world does not have a counterpart. Thus, the acoustics of audio recording sites, even studios, are by *definition disparate. With no standardization at the beginning of the chain, there can be no standardization at the end of the chain either. Thus, "good sound" becomes whatever one wants it to be, or thinks it should be, or tries to make it be. This one difference (film sound's carefully defined standards as opposed to audio's more eclectic approach) has led to many arguments. While it isn't the purpose of this Guide to settle them, you should note that home theater concepts have evolved over many years of experimentation and careful research. SMPTE 202M is only one of the results.*

A QUESTION OF IMAGE

Traditional stereo music systems depend on two separate channels of information to create their sense of depth and imaging. From one point of view, it's a miracle that "stereo" works at all! Think of it for a moment: "soundstaging," at least as we understand this nebulous if highly desired quality, has no real meaning. In fact, one could make a valid argument that soundstaging is but the illusion (in our minds) of a distorted reflection (the recording) of an event (the original concert) we never even attended! And yet, we all admit that two-channel stereo music reproduction is more pleasing in its spatial qualities than is mono (single channel) playback. Whether or not the result is strictly accurate is beside the point — we simply enjoy it more.

In The Beginning, There Were Three . . .

Surprisingly, even industry-standard two-channel "stereo" is something of a compromise. Bell Labs, the telephone company's prestigious "think tank" and birthplace of the transistor, conducted extensive experiments during the '20s and '30s that lead to the development of a three-channel music system with Left, Center, and Right speakers.

This system excelled in creating the sonic illusion of depth. So effective was this approach that Bell Labs conducted a demonstration in 1934 with an orchestra in Constitution Hall in Philadelphia — and Arturo Toscanini standing at the podium of a concert hall in Washington, D.C.!

Rather than facing the orchestra itself, Toscanini stood in front of three speakers hidden behind an acoustically transparent stage-to-ceiling scrim. As the music originating in Philadelphia played through the speakers, he adjusted the relative volumes of each to emulate how he would control orchestral dynamics.

The point of the experiment was to demonstrate the dimensional accuracy possible with this array: Three channels created a solid image that remained relatively constant regardless of where a person sat in the concert hall – exactly as it would if the entire orchestra was actually in the hall! Because of its dimensional accuracy, the Bell researchers soon named this technique "stereophonic."

Notice the similarity between these findings and the film industry's experimentation at roughly the same time: Three-speaker arrays provided substantially better imaging over a much wider area than two-speaker arrays.

Why Only Two?

If all the early research pointed to three-speaker arrays as superior, why did we end up with two-speaker "stereo"? Very simply stated, the LP made us do it. Invented in 1948, the LP provided the first viable method of getting multiple "channels" of sound into the home.

Some of these early LPs were very different from what you might imagine. Rather than one continuous groove, some experimental LPs had two separate grooves on the same side, one beginning at the record's edge and ending midway across the LP, the second beginning there and terminating at the inner retaining groove.

Playback required a "Y" or "V" shaped tonearm with a cartridge on each "branch." The cartridges were spaced so that the first played the outer groove while the second played the inner! As you can imagine, this ungainly arrangement proved very difficult to use in the real world and soon it, along with other equally ingenious but equally awkward approaches, was abandoned.

In 1958, the first commercial "stereo" LP appeared. It was limited to two channels because the "V" shaped record groove had only two walls, each of which carried a separate signal. From this time on, two-channel "stereo" became the norm even though two-channel reproduction carried substantial spatial penalties compared to previous "multi-channel" (i.e., three-channel) approaches.

Just what are the two-channel-only penalties we've all lived with?

The biggest one is that the "sweet spot" (i.e., the listening position which reveals the most accurate and extended image) is very narrow. This tends to make "listening to the stereo" a rather solitary pursuit as the 3 dimensional effects tend to collapse quickly as you move off axis of the center line between the two speakers. Of course, the degree of collapse depends on the speakers themselves, their placement in the room and the acoustical properties of the room itself.

Three-channel arrays almost always present a more dimensional perspective over a much wider listening area. They were also more immune to the effects of acoustically "bad" rooms, whatever their size and shape. Although some have made the seemingly convincing argument that three-channel arrays are best suited to large spaces like movie theaters and do not add to the home experience at all, this is simply not true.

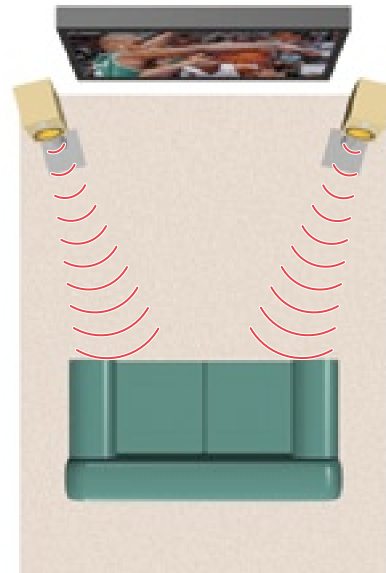
Despite their advantages, three-channel arrays are NOT magically immune to the laws of acoustics. But it does mean that they're more likely to present a convincing sense of dimension regardless of the source.

Another factor in favor of three front channels is that room setup tends to be much less time consuming. We're all too familiar with the seemingly endless search for the "perfect position" that allows two speakers to really bloom with a spatially exciting presentation. Most of this difficulty arises from just that – our use of only two speakers!

The point here is that three speakers, particularly in a home theater system specifically designed to make the best use of a center channel speaker, add a substantial degree of realism a listener can enjoy from almost anywhere in the room. That's flexibility!

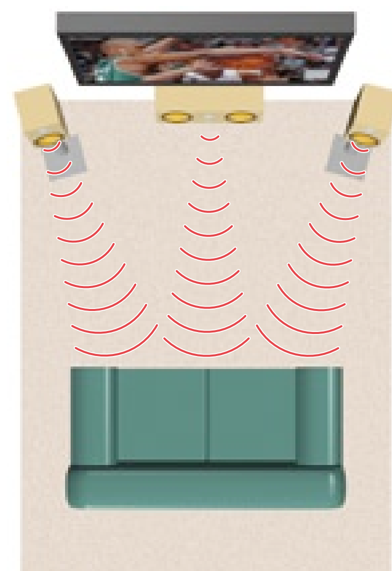
Three-Channel Music Systems?

The three-channel "music only" system is not a new idea. In fact, some audiophiles have been enjoying three-channel reproduction of two-channel material for many decades. These systems use a third speaker located between the right and left speakers just as you would place a center channel speaker in a home theater system. However, the third speaker in a music-only system does not reproduce highly processed "center channel" information but rather just a summed mono signal (L+R) that helps avoid the notorious "hole in the middle" gap sometimes evidenced by poorly set up two-speaker systems.



Widely spaced speakers produce unconvincing panorama with large center gap.

This "hole" phenomenon is an understandable though undesired result of spacing the main speakers too far apart in an attempt to create a convincingly wide soundstage. Rather than present a panorama, these systems usually generate a "here's the left speaker's contribution and over he-e-e-e-r-r-r-e is the right channel information" often referred to as "ping pong" stereo. The third speaker simply melds these occasionally disparate sonic elements together.



Center 'fill' speaker improves image stability and allows wider soundstage without 'hole in the middle' effects.

IS AMBIENCE ENOUGH?

Another difference between conventional two-channel stereo and home theater is that stereo can only hint at ambience while a properly set up home theater system can really envelop you in far more than acoustical "hints."

But before we begin to explain this, let's look at the many different things we mean by "ambience." First, it is that subtle sense of space and air that allows us to distinguish, for example, a cathedral from a nightclub or a forest environment from an open seacoast.

The keys to a good perception of ambience lie in the low-level background sounds that give us acoustic clues that allow us to create the "space" around the main signals. In musical terms, ambience also refers to space, in this case a sense of "air" around the instruments.

With conventional stereo, ambience retrieval is largely a hit or miss affair dependent on the almost-accidental capture of ambience information in the first place. Technically, we define this type of ambience as resulting from "difference" (or L-R) information included in the two main channels.

If we extend the definition of "ambience" to include sounds coming from behind or beside us, we soon find ourselves enveloped in a totally new kind of acoustic environment. Such was the case in the early '70s when some experimenters placed a single speaker behind their listening position and hooked it up to reproduce only that "difference" (L-R) information. At about the same time, the audio industry experimented with various "time delay" devices that compensated for the foreshortening effects of two-channel stereo. The somewhat infamous Quadraphonic Age soon thereafter took that search to places music had never gone before and where consumers, for the most part anyway, refused to follow.

Although these approaches introduced the concept of rear speakers, it wasn't until home theater, specifically the development of Dolby Stereo and its domestic offspring, Dolby Surround, that we were given a real reason to think about the sonic possibilities – and a single standard with which to enjoy them!

The important difference between earlier failures and home theater's success was simple — the movies. Since the late 1970's, movie-goers reveled in the involvement created by "surround" speakers in movie houses. They wanted that excitement for their homes. When the first Dolby Surround-encoded video cassette appeared in 1985, the stage was set for major advances.

For one thing, stereo's "accidental ambience" was replaced with intentional directionality. Now, ambience was not limited to just a sense of acoustic space. Ambience's role grew as directors and sound engineers took advantage of the fact that specific sounds could be placed "back there" for a variety of aesthetic or dramatic reasons.

IN CONCLUSION: THE WINNING COMBINATION

Guided by specifications resulting from years of study and effort, the discipline of film sound has brought some very definite ideas and a welcome sense of continuity to the creation and reproduction of movie soundtracks. Home theater systems combine two highly desirable elements most often lacking in conventional "music only" stereo systems:

1) A **center channel speaker** for increased image stability and enhanced sense of soundstage depth. The center channel generates this perceptible improvement over a much wider portion of the viewing/listening room than is possible with just two front speakers.
2) **Rear "surround" speakers** add much more than a somewhat amorphous sense of ambience thanks to electronics that precisely decode directional information as it was originally intended to be heard.

PREVIEW

In this Section, we'll look at the audio technology that makes home theater possible and give you information to make your shopping and presentations easier and more satisfactory.

We'll begin with the famous Dolby Stereo "matrix" — how it works and what it means in the home. Then we'll explore Dolby Digital – and Dolby Digital EX – to see why and how they were initially developed and what they mean in today's home entertainment world.

Finally, we'll look at Dolby's newest format – Pro Logic II – to see how it fits in the current scheme of things.

INTRODUCTION

We've already seen that home theater systems combine technology from many different areas. Although each contribution is important in its own right, we're going to emphasize once again that it's the sound quality that adds the real emotional impact to movies.

Try this simple experiment: watch one of your favorite movie scenes and turn the sound off half-way through. What do you have left? Just "eye food" — even if the image is truly excellent. And even though the image carries the action, it isn't quite as involving as you thought it would be, is it?

Now, play the same scene but this time turn the picture off. What's left? Well, obviously the soundtrack with its dialog, effects, and music. And that soundtrack captures us in a way a picture alone can't. The sound provokes our emotional response and carries us into the scene even though we can't see it. There are a lot of reasons for this but they all mean **sound quality is vitally important to a good home theater system.**

IT ALL STARTS WITH DOLBY STEREO

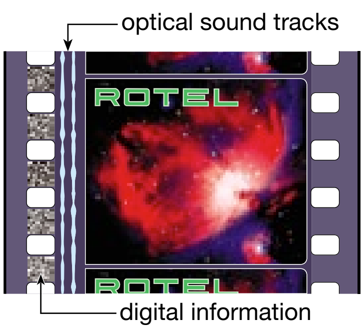
Home theater sound starts with Dolby Stereo, the system developed by Dolby Laboratories to encode four channels of information onto two optical tracks on conventional 35mm movie film.

Since 1976's ***A Star Is Born***, Dolby Stereo has been providing movie-goers with the thrill of involvement as it combines the image stability inherent in three-speaker arrays with separate surround information.

Dolby Stereo squeezes four channels of information (Left, Center, Right, and Surround – or LCRS) from the master mix into two channels (called Lt or Left Total and Rt or Right Total) that are optically encoded onto the film itself.

Dolby Stereo's "active matrix" technology routes sound components as follows:

1) Left channel only information goes only to Lt
2) Right channel only information goes only to Rt
3) Center channel information goes to both Lt and Rt but only after it is reduced -3 dB to avoid unnatural loudness when played back through two speakers
4) Surround signal goes to both Lt and Rt but only after:
    a) Low frequencies below 100 Hz are attenuated
    b) High frequencies about 7 kHz are rolled off
    c) modified Dolby B Noise Reduction increases effective dynamic range
    d) Levels are reduced -3 dB to avoid unnaturally loud reproduction when played back through two speakers
    e) Portion of surround signal sent to Lt is phase rotated +90°
    f) Portion of surround signal sent to Rt is phase rotated –90°



An optical reader in the projector reads the two optical soundtracks (Lt and Rt) on a 35mm Dolby Stereo film. The reader projects a small light through the tracks onto a photocell. The varying track widths control the amount of light passing through the film and, consequently, the photocell's electrical output. These electrical signals are processed by the Dolby Stereo decoder and separate LCRS information is sent to the appropriate amplifier/speaker chain for reproduction.

Dolby Stereo (still used in many theaters as yet unequipped for Dolby Digital) comes in two varieties:

1) "Garden-variety" Dolby Stereo which benefits from Dolby "A" Noise Reduction encoding prior to film mastering and decoding at the theater end to increase effective dynamic range
2) Dolby Stereo SR which uses advanced "SR" Noise Reduction encoding and decoding for even more dynamic range, lower distortion, and more extended response at the frequency extremes than the older "A" technology permitted.

In addition to adding center and surround information in a two-channel format so that we can later extract them for proper playback, Dolby Stereo encoding (regular and SR) also allows complete playback compatibility regardless of the equipment used by the theater.

1) If the theater's sound system in mono, Lt and Rt are simply combined.
2) If the theater has full surround capability, the L, C, R, and S signals are derived from a cinema-grade Pro Logic decoder.

When (principally older) movies are transferred to videotape or DVD, the Dolby Stereo-encoded soundtrack also makes the transition. But because it is now part of a consumer-oriented product, it's called a Dolby Surround soundtrack.

There are several options for playing back this soundtrack in the home and each advance brings us closer to the reality intended by the film's director and sound engineers:

1) We can play the soundtrack back in *mono* with no loss of essential information.

This is basic indeed as mono playback is usually through a TV speaker only. We'll be able to hear something all right but it really doesn't qualify as home theater.

2) We can play it back through a conventional two-channel stereo system.

We'll get a substantial improvement in sound quality as a music system's amplifier and speakers are generally much better than their anemic "TV cousins." In addition, the two- speaker array adds welcome front dimensionality. Although there will still be no surround information as such (because there are no surround speakers, of course), the surround information contained in the Lt and Rt signals will cause the sound to be somewhat more diffuse and enveloping than a conventional stereo-only signal played through the same speakers. This "Home Theater 2.0" approach (so-called because it uses only two speakers), is a common first step towards a full 5.1 home theater system.

3) We can use a Dolby Pro Logic or Pro Logic II decoder to recover a signal for true multi-channel playback.

This is the best way to decode a Dolby Stereo movie soundtrack for home playback. The center channel is enormously effective in producing a solid soundfield to complement the image on the screen. Note that Pro Logic II decoders are more advanced than Pro Logic units and can provide an even more enjoyable experience.

DOLBY STEREO VS. DOLBY SURROUND: IS THERE A DIFFERENCE? AND DOES IT REALLY MATTER TODAY?

At this point, you may be a bit confused by the use of the terms "Dolby Stereo" and "Dolby Surround." And we can't really blame you for not noticing one important difference:

Dolby Stereo describes the two-channel optical soundtrack format for 35mm film.

Dolby Stereo *also* refers to the professional decoders needed to accurately process the soundtrack in the movie theater to restore full-dimensional accuracy. In short, if it deals with film, projectors, or movie theaters in any way, it's Dolby Stereo.

Dolby Surround describes a soundtrack transferred from film to a consumer medium like videotape or DVD (and videodiscs, too, for those who still collect them), and to any TV delivery system. A Dolby Surround soundtrack is identical to a Dolby Stereo soundtrack except that (as we've noted elsewhere) you don't need "A" or "SR" noise reduction to play it back with full accuracy.

Historical note: About four years ago, "Dolby Stereo" magically morphed into just plain "Dolby." That's because lots of confused movie-goers kept asking theater operators why they were just getting "plain old stereo" rather than "surround sound." It's all in the name!

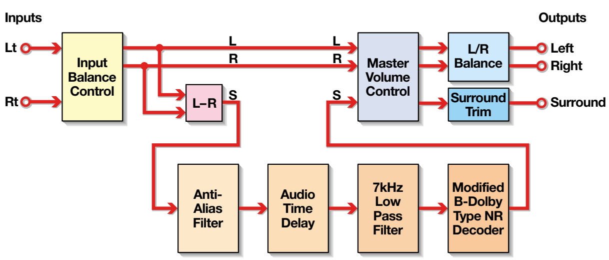
Thus, Dolby Surround is a consumer-only term that means either the encoded soundtrack itself on a tape or disc or the very simple circuitry first introduced to decode it.

DOLBY SURROUND DECODING

Dolby Surround decoding appeared in 1982 and was the first way a consumer could decode a Dolby Stereo soundtrack. It was essentially a three-channel format (LRS) with "passive" decoding for the rear channels.

Dolby Surround's decoding was limited, however:

- Separating front from surround information was not entirely effective. Dolby Surround decoders required manual calibration with a separate input balance control to "null" the L-R information and, if this was not done accurately, front/rear separation could be limited to only 3-6 dB.
- Movement (panning) between front and surround channels was not always smooth.
- There was no provision for a center channel speaker and the "phantom center" provided by normal stereo reproduction of the front channels was only effective for those in the relatively narrow "sweet spot."



Dolby Surround processing derives surround or rear-channel information and leaves L & R information virtually intact.

DOLBY PRO LOGIC DECODING

Dolby Pro Logic, the consumer equivalent of professional Dolby Stereo decoding, was an enhanced method that provided several advantages over Dolby Surround. There were some differences between professional Dolby Stereo and consumer Pro Logic decoding: Pro Logic didn't (and still doesn't) have "A" or "SR" noise reduction because the need for this was already eliminated when the signal was transferred from film to video. Home decoders also use much less surround delay because rooms in homes are so much smaller than theaters. However, in all other ways, Pro Logic decoding duplicated the accuracy of Dolby's best professional decoders of the day.

Introduced in 1987, Pro Logic gradually became the home theater standard. Although most early decoders employed analog circuits, a few took advantage of digital processing right away. Many others followed.

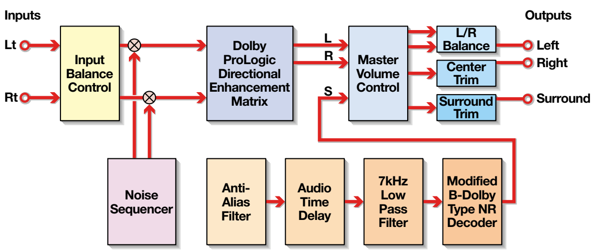
Pro Logic's advantages included:

- Provision for a center channel speaker to anchor dialog to the screen for better dimensional stability

- Better separation between LCR and S channels (at least 25 dB) for more accurate ambience presentation
- Smoother transitions (or "pans") as the sound moves from front to rear
- Potential for digital implementation to further enhance accuracy and consistency

The most significant Pro Logic circuit block is the one labeled Directional Enhancement Circuitry (DEC). Sometimes called Active Steering, or Active Steering Logic (hence the name Pro *Logic*), DEC continuously monitors the signal to determine which directional element is dominant at any moment and then proportionally enhances that dominance to more accurately anchor the image in a three-dimensional soundfield.

For example, if a voice is slightly louder in Rt than in Lt (everything else being equal), Active Steering Logic will slightly boost R output for as long as the directional difference exists. If the voice is loud enough to make Rt significantly stronger than Lt, the DEC block will boost the R output even more. If the voice is equally strong in both Rt and Lt, the DEC will increase C output for as long as necessary.



This Dolby Pro Logic block diagram is remarkably similar to Dolby Surround processing with two significant additions: (1) Directional Enhancement Circuitry that dramatically increases separation and (2) a Test Tone Noise Sequencer which makes setup much easier and more accurate.

DOLBY SURROUND/PRO LOGIC DECODING AND TIME DELAY

One of the most misunderstood components of Dolby Surround and Dolby Pro Logic decoding is the time delay applied to the surround (or rear channel) signal. To better understand why time delay is so important, we need to look at the way we perceive sounds, particularly at how we discriminate against reflections that would otherwise confuse our sense of sonic direction.

Let's start with the fact that neither Dolby Surround nor Pro Logic decoding were perfect. Of particular interest is the fact that they both "leak" some front channel information into the rear or surround channel. If we played the rear channel information back without delay, the near-simultaneous arrival of the common elements (particularly speech or dialog) from front and rear would totally confuse our sense of aural direction. We'd hear dialog, for example, coming from some ill-defined point in the viewing/listening room. In severe cases, the dialog might first appear to be coming from the front, then suddenly "pop out" from the rear before collapsing into the front again. Regardless of the severity of this effect, the fact that it could happen at all would be very disconcerting and distracting.

To avoid this, Dolby specified delay for the rear channels. By postponing the arrival of rear channel information, delay gives our ear/brain combination adequate time to "lock in" to the apparent source of a sound coming from the front before any similar "leakage" signal from the rear totally confuses things.

Dolby reduces the chance of this interference even further by imposing a high-frequency filter on the rear channels primarily to reduce the perceptual impact of consonant sounds in speech or sibilant leakage caused by imperfect phase coherency in the Lt/Rt delivery channels.

In other words, the right amount of delay effectively "masks" the arrival of distracting rear channel signals in that the brain simply ignores sounds it has already identified and assigned directional pointers too.

This isn't to say that *all* rear channel sounds are masked: They most definitely are not. Masking works only for sounds that are common to both front and rear (i.e., those that "leaked" during decoding) and has no effect whatsoever on the difference or ambience information that makes up special rear channel effects. Dolby specifies a range of delay times to allow Dolby Surround and Pro Logic decoders to work effectively in a large variety of rooms.

Note: Don't forget that Dolby Pro Logic decoders produce only ONE surround channel output! That might come as a surprise as almost every home theater system has two speakers to reproduce surround information. However, Dolby Pro Logic decoders feed exactly the same signal to both! Actually, you could drive two speakers with one rear channel amplifier and, assuming there was no gross impedance mismatch between the amp and speakers, you'd never hear the difference! (Obviously, that isn't the case today. Indeed, Dolby's new Pro Logic II circuitry provides two separate surround channels.)

As you'll soon see, THX processors can modify even a mono surround signal to create a greater sense of ambience. That's why a THX system always needs two separate amplifiers for surround information.

DOLBY DIGITAL

Dolby Digital (first called Dolby AC-3) is the most prominent example of digital data transmission for audio applications. It is the *de facto* standard for DVDs worldwide, the mandated standard for DTV in the U.S., and has gathered a lot of support in the computer gaming industry.

Originally developed for DTV applications, Dolby Digital made its commercial debut in 1992 with the release of *Batman Returns*. (The delay in adopting a DTV standard caused Dolby Digital to make its first appearance as a film format.)

A major reason for the format's success was that Dolby succeeded in placing blocks of data between the film's sprocket holes. Although the available real estate was small, Dolby succeeded in producing a bit rate of 320 kilobits per second. Dolby Digital films also carried the more common Dolby Stereo (or just plain "Dolby") optical soundtracks, too, so one film print version could be used anywhere.

Dolby Digital's consumer-oriented variant provides several advantages over earlier Pro Logic technology.

1) Dolby Digital is, not surprisingly, *digital* in nature as opposed to Pro Logic which is essentially analog (even though decoding can take place in the digital domain.)

2) *Dolby Digital can (repeat – **CAN**) carry six discrete channels of information* as opposed to Pro Logic's four-channel matrix. This has significant impact on the dimensional clarity of the resulting soundfield.
   a) *Left and Right surround information is totally separate* as opposed to Pro Logic's mono surround and can be full bandwidth (20 Hz - 20 kHz) compared to Pro Logic's 7 kHz limitation. In addition to providing more lifelike ambience information to a movie soundtrack, this also makes Dolby Digital surprisingly suitable for multi-channel music reproduction.
   b) A fully independent LFE (Low Frequency Effects) channel carries all necessary bass information for maximum impact and dynamic range.

How Dolby Digital Works

Dolby Digital addresses a major problem in conventional digital data structures like Pulse Code Modulation (PCM). PCM data streams are very dense. A two-channel PCM signal of industry-standard 16-bit words at 44.1 kHz sampling rate requires a data flow in excess of 1.4 million (mega-) bits per second just for the basic audio data! And that doesn't even include the "redundancy bits" needed for error correction or the additional flags and signals that keep the data flowing in an orderly manner!

Dolby Digital allows economic storage and transfer of audio information by using far fewer bits than PCM. The trick is in determining which bits you can "throw away" (for a less demanding data stream) while still maintaining a high degree of signal integrity.

The key to Dolby Digital's performance (indeed, the key to any data compression system for audio) lies in psychoacoustics, particularly a human hearing phenomenon called *auditory masking*. Simply stated, auditory masking means that the ear/brain combination focuses on intelligible data even when there is a lot of random data (noise) very close in frequency. (A good example of auditory masking in action is your ability to hear whispered conversation in a noisy bar.)

Dolby Digital divides the audio spectrum of each channel into as many as 256 narrow frequency bands keyed to the selectivity characteristics of human hearing. This enables a series of sharp filters to keep the low bit system's quantization noise very close in frequency to the audio data. The system then reduces or eliminates this noise whenever there is no audio signal to mask it and thus preserves subjective sound quality.

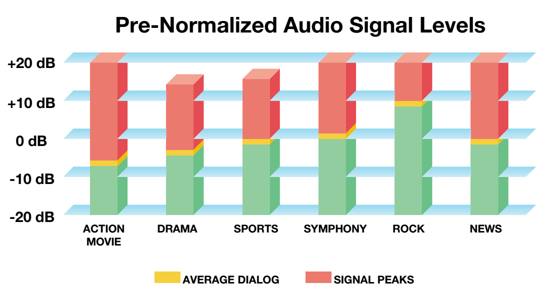
Dolby Digital uses two interdependent techniques:

1) "Smart filters" ensure that a sufficient number of bits are used to describe the audio signal in each band, thus thoroughly masking the noise inherent in "low bit" systems.
2) Dynamic "bit allocation" strategies ensure that channels with significant audio information benefit from a larger percentage of available bits from a common pool. This maintains signal integrity even under demanding conditions.

The consumer version of Dolby Digital began life with a bit rate capability of up to 640 kilobits per second (kbps) but was first used at 384 kpbs for 5.1 channel sources on laserdisc. This is about a fourth of the data rate needed for just two channels of PCM audio. Today, most new DVDs have Dolby Digital tracks at the 448 kbps rate, the same rate possible for DTV audio.

Added Benefits

Dialog Normalization

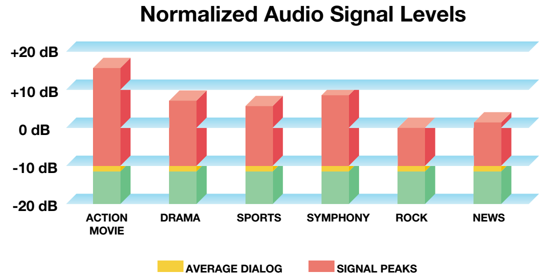


This shows normal and peak levels for common program types. Note the wide variation.

Different video sources usually produce different audio levels as well as different images. That's one of the major annoyances we encounter as we "channel hop" or go from TV to VCR to DVD player. This results from the various ways engineers mix soundtracks and from the different production standards in place for various program types. We can see these differences very clearly if we look at the differences between average dialog levels and program peaks in various types of programming.

If we look at "Action Movie" and "Rock," both program types exhibit the same peak levels. But "Rock's" dialog (or "average") level is much louder. That means you'll hear a substantial and disconcerting jump in perceived loudness when switching from a movie to a concert and might want to reduce your system's volume setting to a more comfortable level.

To avoid this problem, Dolby Digital includes something called "Dialog Normalization" to analyze average to peak level differences and "normalize" the overall playback volume level so that the perceived level shift is not nearly as annoying as it might be otherwise. The reason this works is simple. When we listen to a movie or a TV show, we typically set the volume based on that program's dialog level. If it's too loud, the program "shouts" at us. When it's too quiet, we can't make out the story line. Dolby uses the term "dialog normalization" in this light. It shouldn't be confused with somehow only adjusting the dialog portion of the mix separately from the rest of the information contained in a source's soundtrack.



Dialog Normalization evens out program-to-program differences.

Notice that overall dynamic range (the difference between very soft and very loud sounds) remains the same. That means dynamic contrasts, one of the most important aspects of lifelike sound, are completely unaffected. Incidentally, the amount of level change is preprogrammed by the studio engineer preparing the Dolby Digital bitstream, and it remains constant for the entire program.

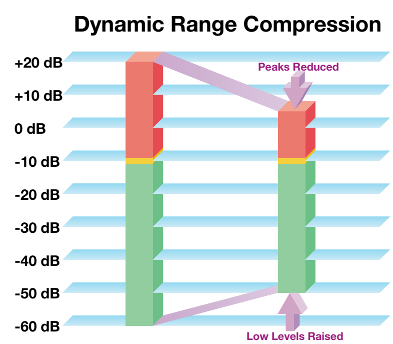
"Late Night" Compression

This feature *does* affect dynamic range but the effect is controllable and often desirable when, for example, you're watching *Godzilla Gets Down With King Kong* late at night.

As we just saw in the previous section on Dialog Normalization, an action movie has very high peak levels – usually an explosion! When we watch late at night and turn the system volume up high enough to hear soft whispered dialog, we'll probably be blown out of our chair when the bad guy gets it. .

If *you're* half-asleep when this happens, you'll just levitate. At least momentarily. But if there's a baby sleeping in the next room, you'll be up for the next three hours or so – and it won't be to watch the movie either!

So Dolby Digital provides a (usually adjustable) dynamic range control that raises low levels so you can hear soft sounds without raising overall system volume. It also reduces peak levels so you won't be walking the floor with your favorite screaming kid once the bad guys get blown to smithereens!



Dolby Digital's adjustable dynamic compression raises soft sounds and lowers loud ones so you can watch all you want without disturbing people around you.

Perhaps surprisingly, this compression isn't accomplished in the Dolby Digital decoder itself. Instead, the production team responsible for whatever program you're watching (usually the director and sound engineers) decides what gets compressed and by how much. This information is coded into the data stream. (Technically, these instructions are part of what Dolby refers to as "metadata.") This means no decoder-to-decoder variability. However, some decoders allow you only one choice – "on" or "off." Others allow different levels of compression provided, of course, that the soundtrack has been so encoded. In either case, you can't get more compression than the mix team allows!

Bass Management

Integrating a subwoofer's output with those of five full-range loudspeakers can be a real problem.

In the Age of Stereo, many people added "subs" in hopes of increasing bass output, particularly when the main speakers were a bit anemic. But some systems just sounded bass heavy or suffered from sonic discontinuities in the "transition zone" between subwoofer and full-range speaker.

That's why Dolby Digital includes "Bass Management."

Simply put, Bass Management defines which speakers reproduce low frequencies. To do this properly, the filters in the Bass Management circuit need to "know" the low frequency capabilities of a system's main speakers.

Once that information is programmed in during initial setup and calibration, Bass Management uses that information to optimize system performance.

Here are three examples.

1) When a system's main (Left and Right) speakers are somewhat bass-shy ("satellite" speakers, for example), Bass Management filters out the low frequencies from all channels and directs them to the subwoofer. Even if your speakers are capable of reasonably deep bass, a good subwoofer may be even better. In that case, you may want to experiment a bit to see which mode (a full range signal going to each speaker with the sub getting only an LFE signal, or directing *all* the bass to the sub) gives you the most convincing in-room impact.

2) When a system's main speakers have serious low frequency capability, Bass Management routes all the "subterranean rumblings" from all six channels to the main speakers. (This is particularly useful for those systems that do *not* have a subwoofer.)

3) If all the system's speakers are full range and thus bass-capable, Bass Management directs the low frequency information accordingly.

If this option is chosen, every speaker receives full range audio. The LFE information is sent to the subwoofer only if one is used – and one usually is. No subwoofer? Then "LFE bass" will usually be sent to the main L/R speakers, but the manufacturer may offer other options as well.

Outboard processors must offer at least these three options. (Others are allowed but they're at the discretion of the manufacturer.) Dolby Digital-capable receivers must include at least Options 1 and 2 above.

There's also some flexibility as far as crossover frequency is concerned. Although Dolby recommends 80 Hz, any frequency between 80 and 120 Hz is allowed. (Others may be included at the manufacturer's option.) The minimum crossover slope is 12 dB/octave low pass (for the subwoofer) and 6 dB/octave high pass (for the full range speakers).

Time Alignment

Think back to the discussion of "delay" in the section on Dolby Pro Logic. We're about to revisit it. As you remember, a proper delay setting offsets the effects of front-rear "bleeding" – an inherent artifact of Dolby Pro Logic decoding.

Although Dolby Digital does not exhibit "bleeding" as such due to its discrete nature, we still need delay. There are three reasons:

(1) Home theater systems perform differently in different rooms.
(2) We will still play Dolby Surround encoded material.
(3) There are still panned effects in 5.1 programs that need to arrive in the proper time relationships.

Time Alignment addresses these situations. It is required in all Dolby Digital processors but comes in various flavors.

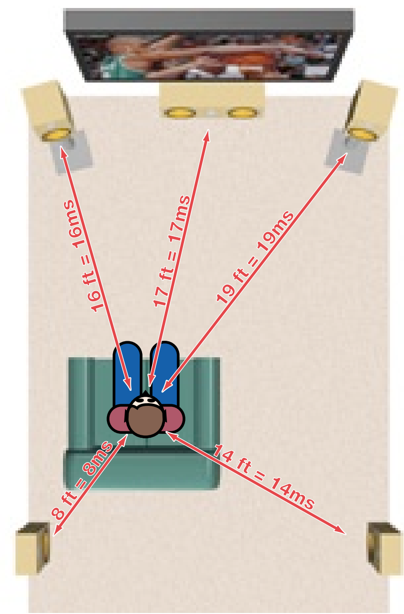
Before we dive into the details, remember that all Dolby Digital decoders provide some form of system calibration for surround delay. Once the decoder is adjusted for one format - Dolby Pro Logic or Dolby Digital - it is automatically set for the other.

Time Alignment makes this process simpler by asking us to tell the processor how far from our favorite viewing/listening position our speakers are.

Basic Time Alignment adjusts only center and surround channel delay times. It uses the main - L & R - channel distances as a reference.

More advanced Time Alignment implementations allow you to enter all measurements individually so the processor can automatically set the delays for all speakers.

In some cases, the processor may ask you to enter the surround delay in milliseconds. To figure out this number, simply subtract the distance in feet from the listener's head to the surround speakers from the distance between the listener's head and the front left or right speaker. This gives you the proper delay in milliseconds.



After you enter distance measurement, Time Alignment automatically calculates delay settings.

A NOTE ON THE SPEED OF SOUND

We usually refer to the speed of sound as 1,088 feet per second. Of course, that's at sea level in a low-humidity atmosphere at 32° F. Changes in altitude, temperature, relative humidity, and other factors change this speed slightly.

An acceptable equivalent is 1,000 feet per second, or 1 foot per millisecond. Remember that figure as we'll see it again and again.

Dolby Digital Does Dolby Pro Logic

It was important that Dolby Digital products also handle the large base of existing Dolby Surround content so Dolby Laboratories made Pro Logic decoding a mandatory part of the Dolby Digital decoder spec. In practical terms, this means that every Dolby Digital-capable processor will also decode Dolby Surround sources. That's an important consideration for those with extensive videotape collections. Remember that there are literally thousands of prerecorded VHS tapes that may never be released on DVD or any other Dolby Digital-capable format. Added to that is the continuous flow of Dolby Surround movies and other programs on television, cable and satellite. For people who value these programs, Pro Logic lives!

Almost all Dolby Digital processor ICs perform Pro Logic decoding in the digital domain. That means all such Dolby Digital components must include analog-to-digital converters so an analog Lt/Rt audio signal from a VCR, for example, would be available in digital form for the IC to crunch.

Downmixing

Downmixing is another Dolby Digital advantage. This refers to a Dolby Digital decoder's ability to generate a stereo or even a mono output when the source is really 5.1 multi-channel.

This can provide a significant advantage in playback flexibility.

For example, you could play a full 5.1 Dolby Digital source through a Dolby Digital decoder and get all the dimensionality the creative team wanted you to hear.

If you had just a Dolby Pro Logic decoder, you'd still hear everything in the mix just as well as with regular Dolby Surround encoded material. But directionality would suffer a bit compared to 5.1 discrete playback.

Playing that same source through a conventional two-channel stereo system would still bring you full-bandwidth sound – minus, of course, those specific directional clues normally available from center and surround speakers.

In terms of hardware compatibility, downmixing means that a DVD player can work with a stereo TV or stereo music system, a Dolby Pro Logic system, or a Dolby Digital (Surround EX included) multi-channel system. Remember that all DVD players have two (L and R) analog outputs. Now you know where those signals come from!

Are "Dolby Digital" and "5.1" Synonymous?

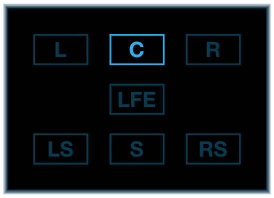
In a word, NO!

Although it's easy to fall into the trap of thinking that Dolby Digital automatically delivers 5.1 channels of sound, that's absolutely incorrect.

A Dolby Digital soundtrack could be mono, stereo, Dolby Surround-encoded. Finally, it can be (*can* be!) true discrete 5.1 channel sound. In fact, if it's Surround EX-encoded, it would be a "5.1+" source, wouldn't it?

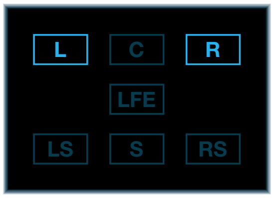
In order to help clear up any misconceptions, Dolby has suggested that their hardware licensees provide a display on Dolby Digital-capable components that tells the user how many channels the Dolby Digital soundtrack is actually providing.

The graphic display for hardware is almost self-explanatory. Here are the variations it provides.



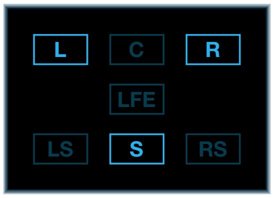
Mono

*This shows a Dolby Digital-encoded **mono** soundtrack.*



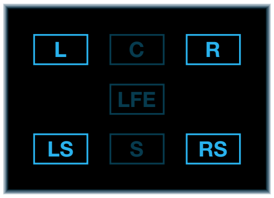
Stereo

This indicates a normal stereo signal.



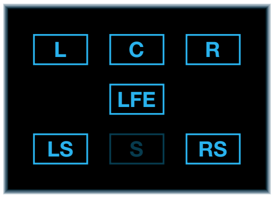
Dolby Surround

Here's your Dolby Surround-encoded soundtrack.



Quad

The Dolby Digital datastream is carrying four different channels of sound.



5.1

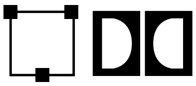
All the channels of your home theater system are receiving discrete information.

There's also a symbol structure for software so you can see how many channels of information are on the soundtrack even before you play the source.

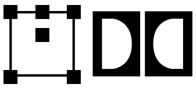
The software diagrams perform the same function as the hardware display. Used together, they give you a quick way to check that all system settings are correctly adjusted so that you can enjoy everything the source's soundtrack has to offer.

Mono

Stereo



Dolby Surround



5.1 Channel Dolby Digital

WHAT IS "5.1" . . . AND "6.1" . . . AND "7.1" . . . AND SO FORTH?

With the introduction of new audio formats like Pro Logic II, Surround EX, and DTS ES Discrete, we need to take a closer look at something that confuses even veteran audio salespeople – the mysterious "5.1" (and now "6.1" and even "7.1") designations. This is particularly important as we move from the world of engineering descriptions to marketing excess.

The chart below is part of our effort to simplify the situation. You'll note that we've added a "5.1+" designation to the usual roster of "x.1" descriptions. We haven't done this lightly but we do think it's far more accurate and thus more descriptive than some of the other terms you'll see thrown around today. The reason we did this was in recognition of the fact that the "5" in "5.1" originally referred to five totally discrete channels of information.

Case in point is Surround EX, whether from THX or Dolby. In fact, Surround EX, as we explain elsewhere, makes use of a matrix technique to "piggyback" a Center *surround channel onto the two discrete surround channels. This Center surround channel, then, is NOT discrete and calling Surround EX (or DTS ES) a "6.1" system is simply misleading.*

To put all this in perspective, we need to begin with mono (1 channel). After migrating to stereo (2 channels), we journeyed through full surround (4 channels), and are now well into the age of 6 and 8 channel audio. The chart outlines some of the differences.

As you look it over, remember that the original and still-confusing "5.1" designation came about because movie-based digital multi-channel formats began with the idea of five full range (i.e., capable of delivering the entire audible spectrum from 20 Hz to 20 kHz) channels: Left Front, Center Front, Right Front, Left Surround, and Right Surround and one LFE (or Low Frequency Effects) "quake-certified" channel strictly for bass information.

This bass channel was limited to frequencies below 120 Hz (or less than one-tenth the usual range of 20 Hz - 20 kHz) and became known at the ".1" channel in a "5.1" channel system. It was just a continuation of the same confusing silliness that brought us so-called "6.1" and "7.1" systems, too!

Name Of Format	***Number of Channels***	***Frequency Response***
Mono (Analog)	1 (discrete by definition)	Full range
Stereo (Analog)	2 (discrete)	L(eft) & R(ight): Full range
Dolby Surround (Analog)	3 (matrix)	L & R: Full range S(urround) (mono): Restricted bandwidth No LFE (Low Frequency Effects)
Dolby Pro Logic (Analog or digital)	4 (matrix)	L, C(enter), & R: Full range S (mono): Restricted bandwidth No LFE
Dolby Pro Logic II (Analog or digital) (Description also applies to DTS' Neo:6 matrix)	5 (matrix)	L, C, & R: Full range LS & RS (stereo): Full range No LFE directly from the decoder itself although Pro Logic II-capable processors and receivers may use Bass Management to derive one.
"5.1" (Digital) (Dolby Digital, DTS, etc.)	6 (discrete)	L, C, & R: Full range LS & RS (stereo): Full range LFE: Restricted bandwidth

"5.1+" (Digital) (THX Surround EX, Dolby Surround EX, DTS ES)	7 (6 discrete + 1 matrix)	L, C, & R: Full range LS & RS (stereo): Full range CS (Center surround): Full Range LFE: Restricted bandwidth
"6.0" (Digital) (DVD-A, SACD, Chesky, DTS Neo:6)	6 (discrete) DTS Neo:6 is matrix	*Normal DVD-A and SACD:* L, C, & R: Full range LS & RS: Full range No mandated LFE but additional channel may be used and configured as desired. *Chesky variant:* L & R: Full range LH(eight) & RH: Full Range LS & RS: Full range No LFE *Neo:6 Matrix* L, C, & R: Full range LS, CS (Center Surround), & RS: Full range No LFE
"6.1" (Digital) (DTS ES Discrete)	7 (discrete)	L, C, & R: Full range LS & RS (stereo): Full range CS (Center surround): Full Range LFE: Restricted bandwidth
"7.1" (Digital) (THX Ultra2 spec)	8 (6 discrete, 2 derived from combination of LS, RS, and matrixed Center surround)	L, C, & R: Full range LS & RS (stereo): Full range CSL and CSR (Center surround Left and Right): Full Range LFE: Restricted bandwidth
"10.2" (Digital) (Proposal by Tom Holman)	12 (discrete)	L, C, & R: Full range LH & RH: Full range LW(ide) & RW: Full range LS & RS (stereo): Full range CB: Full Range LFEL and LFER: Restricted bandwidth

Note that there is quite a bit of dissention over the "5.1" format for music use. Some very knowledgeable people feel that even 6 channels is not enough to reproduce the true ambience of music performed in an enclosed space. For example, Michael Gerzon, a noted British mathematician and acoustician, once stated that 32 loudspeakers would be a reasonable way to approximate a real soundspace. As correct as he may have been, obvious questions of practicality (both space and cost) must intrude on the theoretical.

In addition, many musicians, techno-savvy producers, and engineers are not at all happy with the way a center channel speaker interferes with the image created by the main (L & R) speakers, particularly when the center channel speaker is not properly integrated with the rest of the system. (This happens far more often than we might want to admit.) This dissatisfaction, by the way, was the impetus for David Chesky's "6.0" system described in the table above.

Although there seems little chance that these aesthetic concerns will triumph over market realities, you should be aware that we may yet see new formats or conventional formats with revised channel allocations a la Chesky. In fact, the limitations of 5.1 convinced Tom Holman of THX fame to develop his "10.2" system!

STEREO OR SURROUND FOR MUSIC?

At heart, this debate is largely an aesthetic one. Multi-channel advocates argue that two channels simply aren't sufficient for music. Traditionalists state with equal fervor that five or more channels is a temptation that overenthusiastic or underexperienced producers and engineers simply can't resist as they explore new perspectives that simply don't echo the way we experience music in real life.

These are not, by the way, necessarily conflicting positions. Those who support multi-channel music also don't want to see poor practitioners making a mess of it! At Rotel, we're more sympathetic to the traditional two-channel view. Among other considerations, we know that audiences usually don't sit surrounded by musicians and that the usual multi-channel perspective (one that puts us in the middle of the music) is an unnatural one.

However, we also know that "surround music" is a very young art form and we have confidence that those who make the music will quickly learn how to use surround's potential.

As an aside, remember that although multi-channel proponents argue vociferously that two channels don't cut it, some equally knowledgeable acousticians argue quite convincingly that even 5.1 channels aren't enough.

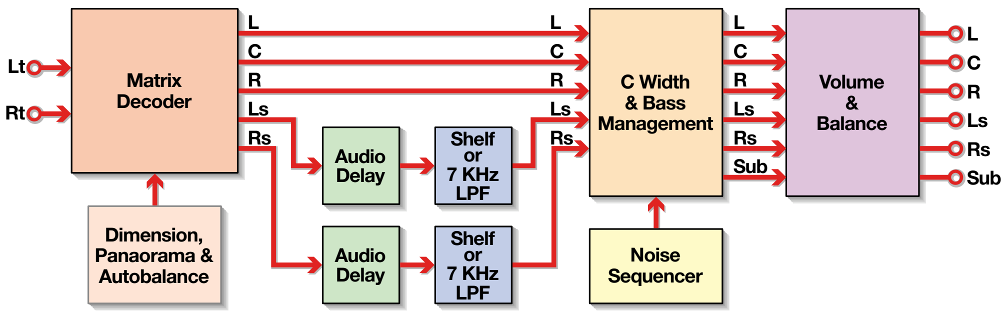
Suffice it to say that these questions are not yet fully answered and are not likely to be for some time. Just as it took us a while to learn how to use "stereo" effectively, the music industry needs to learn new techniques to best take advantage of surround's spatial potential.

Notice, however, that we haven't questioned the aesthetics of the movie experience. That's because multi-channel movie sound reproduced through five or more full-range and a subwoofer channel is the standard today.

DOLBY PRO LOGIC II

Some might question placing Dolby Pro Logic II at the end of this section. After all, isn't it just an updating of the much older Dolby Pro Logic analog processing? The answer to that is "Yes." And "No."

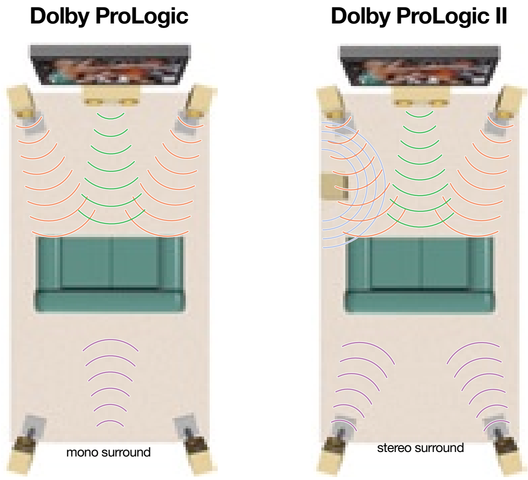
Although Pro Logic II is compatible with the same Dolby Surround programs used with Pro Logic decoding all these years, it is far more refined. Initially developed by Jim Fosgate, one of the industry's true "matrix masters," Pro Logic II brings significantly new capabilities to enhance the viewing/listening experience.
Although a manufacturer can implement Pro Logic II decoding with digital circuitry, the process is essentially analog in nature. It works in a manner that is similar to Pro Logic but far more sophisticated.



Here's a simplified view of the individual stages in a PL II decoder. You'll find explanations for all these stages in the tables and paragraphs that follow.

The impetus for PL II (as we'll call it from here on out) was Dolby Digital. As you'll remember, Dolby Digital provided many sonic advantages over Pro Logic. The only problem with Dolby Digital (at least the only problem germane to *this* discussion) is the fact that many sources – videotape, TV, and FM broadcasts, etc. – can't deliver Dolby Digital's advantages. Although Pro Logic could still be used, it suffered in comparison with Dolby Digital. PL II addresses these problems.

First, PL II offers stereo surround information. (Pro Logic's surround output, as you remember, was mono.) In addition, Pro Logic's mono surround track was bandwidth-limited (no output above 7 kHz) while PLII's stereo surround channels cover the entire audible spectrum for more natural reproduction.



Pro Logic II brings true stereo surround.

Another big plus for PL II is that you can optimize it for either movies or music. Some PL II decoders provide further adjustments. The following table shows some of the possibilities.

Feature	***Pro Logic***	***Pro Logic II***
Source material	Dolby Surround encoded programs	Dolby Surround encoded programs Stereo music programs
Output Modes	"3/1" surround (LCR & S) "2/1" surround (phantom center) "3/0" (Dolby 3 Stereo)	"3/2" surround (LCR; LS, RS) "2/2" surround (phantom center) "3/0" (Dolby 3 Stereo) *Optional* Pro Logic emulation modes: "3/1" surround (LCR & S with 7 kHz surround limitation) "2/1" surround (phantom center and restricted surround bandwidth)
Surround Channel Bandwidth	100 Hz - 7 kHz only	Full range
"Dimension" control	No	Yes Adjusts apparent "front/rear" positioning of sonic image. Optional for home and automotive products. May be called "front/rear fader" for car applications.
"Panorama" control	No	Yes Increases apparent width of front image. Optional for home and automotive products.
"Center Image Width" control	No	Yes Allows producer/engineer to better integrate vocals across soundstage.
Custom Mode options	No	Yes Standard Movie Music Pro Logic emulation Matrix (passive) as pre-processing for Dolby Headphone

Some of the points in the above table need further explanation.

Source Material: Notice that PL II can be used with conventional two-channel music sources in addition to Dolby Surround-encoded sources.

Although you can use Pro Logic with music, the results are random in their effectiveness and almost never coincide with the artists' intentions. PL II's more sophisticated circuitry adds coherent spatial qualities to most musical sources.

Output Modes: Don't confuse this with "movie" or "music" modes. These simply indicate the number of outputs you can derive from a two-channel input signal depending on your system configuration and preferences.

Although the biggest difference between Pro Logic and PL II appears to be stereo surround information, careful listening will reveal other substantial differences.

Surround Channel Bandwidth: Self-explanatory. Remember that Pro Logic's surround output was bandwidth limited to guard against the audible effects of "leakage" from front to rear that would prove distractive. PL II's more sophisticated steering circuitry avoids this problem altogether.

"Dimension" control: This is an optional control for both home and automotive PL II decoders but will probably appear more often in car units. It allows the user to gradually adjust the soundfield either towards the front or the rear of the listening environment. It can also help restore a more natural front rear balance with certain recordings.

"Panorama" Control: This extends the front stereo image to the surround speakers to generate a "wraparound" effect to complement certain sources.

Center Image Width Control: This adjusts the center image so that it is heard

1) only from a center speaker

2) only from the L & R speakers as a "phantom" center
3) from various combinations of all three front speakers.

The control lets both driver and passengers in an automobile, for example, enjoy a realistic Left/Center/Right presentation. It improves blending of LCR speakers in a home environment for more natural rendering of vocals.

Custom Mode:

The most important of these is Music Mode. Here, surround channel high frequencies are slightly attenuated via a "shelf filter" to approximate what happens to reflected sound in a concert hall environment and to minimize distraction. If the center and surround speakers are closer to the viewing/listening position than the main L & R speakers, the PL II decoder automatically compensates to insure *coincident arrival* of all signals. This prevents unwanted sonic smearing as the sounds from all speakers arrive at the listener's ears. Autobalance, a circuit retained from Pro Logic and still used in PL II's Movie mode to insure proper center channel output, is turned off to make sure that intentional off-center instrumental or vocal placement is retained.

Here's a short comparison of the differences between PL II's Movie and Music modes.

Feature	**PL II – Movie**	**PL II – Music**
Surround Filter	None used	High frequency shelf
Surround Delay	Yes (10 milliseconds)	None
Panorama	Off	Optional
Dimension	Off	Optional
Center Width	Off	Optional
Autobalance	On	Off

PREVIEW

When it first appeared, DTS promised improved sound quality for films and, later, for consumer products like CDs and DVDs. In truth, DTS' much-vaunted higher bit rates have many theoretical advantages.

DTS initially stunned theater-goers with the shudderingly effective Jurassic Park soundtrack and, for some time, was the preferred choice as movie theaters moved to install high-quality digital multi-channel sound capability.

From there, DTS moved to the consumer market where it briefly challenged Dolby for dominance. More recently, DTS has emerged as a "niche player," catering to those who opt for its claimed sonic advantages.

To that end, DTS has shown several new surround formats that we'll outline towards the end of this section.

In addition to Dolby Digital, DTS – short for Digital Theater Systems, the company that developed and is promoting it – has established itself as an important multi-channel audio format.

FROM THE FILM WORLD . . .

Most of us first heard a DTS soundtrack when we saw Steven Spielberg's 1993 mega-hit *Jurassic Park* at our neighborhood theater.

In contrast to Dolby Digital, which placed the audio information between the sprocket holes of a movie film, DTS' approach put the sound on a CD-ROM disc. The film itself carried only a printed time code which the CD-ROM player synched to. Theater owners saw several advantages in the DTS approach. For one thing, it was less expensive to implement as the CD-ROM player was comparatively cheap in comparison to the optical reader in the projector needed to scan Dolby Digital audio data.

This dual media system (film for the visual image, CD-ROM for the audio data) promised much. For one thing, CD-ROMs have a substantially larger storage capacity than the small "between the sprocket holes" area where Dolby Digital audio data is stored. That meant a DTS soundtrack can have a much higher bit rate than Dolby Digital – 882 kilobits/second compared to Dolby Digital's film-standard 320. This, in turn, meant less compression: 4:1 compared to Dolby Digital's nominal 12-13:1 rating.

Other DTS advantages included the ability to easily offer soundtracks in different languages by simply substituting a different CD-ROM disc. (The film print was the same for all countries.) And the time code markers needed to synchronize sound to image were more robust than the somewhat exposed Dolby Digital audio data.

. . . TO THE HOME ARENA

Soon thereafter, DTS began to promote a modified version of its coding scheme for multi-channel music-only use and, later, as a proposed standard for DVD movies. Now, DTS is an official part of the DVD specification but as an *alternative* choice. (Dolby Digital or PCM audio is now required on all DVDs.)

TECHNICAL OVERVIEW

DTS is not just a "one size fits all" approach to the problems of providing high-resolution digital audio in entertainment formats where restricted "real estate" limits the amount of data. Although that stands behind all DTS applications, some members of the DTS "family" have slightly different data structures.

We can understand this better by reading DTS' own design objectives:

> *The DTS Coherent Acoustics compression algorithm was designed from the outset to perform at studio-quality levels, i.e., 'better than CD', and in a multi-channel format was intended to facilitate a major advance in the quality of audio reproduction in the home in terms of fidelity and sound stage imagery. Another primary objective was that the compression algorithm should be broadly applicable and therefore flexible. Multimedia applications have restricted data bandwidths and therefore demand a 5.1 channel mode operating at 384 kbps or less.*
>
> *Professional music applications involve higher sampling rates, longer word lengths, multiple discrete audio channels, and increasingly demand lossless compression. All of these features have been accommodated in Coherent Acoustics.*
>
> *The final important objective was to ensure that the universal decoder algorithm was relatively simple, and future-proofed. This would ensure cost-effective consumer decoding hardware today, and yet allow consumers to benefit from any future improvements realized at the encoding stage.*
>
> From an ***Audio Engineering Society*** preprint distributed at the 100th AES Convention, Copenhagen, 1996 (Workshop 4a-3: *Multi-channel Sound Production Techniques and Technologies*)

In other words, DTS is a kit rather than a single tool. That's because DTS is both "hierarchical" and "scaleable." This means that engineers can define exactly what sonic attributes are particularly important for a project and configure DTS encoding accordingly.

DTS' "scaleability" lets these same engineers configure DTS' variable compression for the least compromised sound in whatever space is available to hold it. Here are some of the choices DTS makes available:

- 1 to 8 channels of multiplexed audio
- sampling rates from 8kHz to 192kHz
- 16 to 24-bit audio equivalent PCM word lengths
- compression ratios from 1:1 to 40:1
- lossless coding mode (variable data rate)
- encoder output data range 32 kbit/s to 4.096 Mbit/s
- linear PCM decoding mode
- down-mixing from n coded channels to n-1, n-2,…etc. output channels
- down-mixing from 5.1 discrete channels to stereo L_T, R_T
- embedded dynamic range control
- re-equalization of all channels independently
- sample-accurate synchronization of audio to eternal video signals
- embedded time stamp and user data
- future-proof decoder

Although all these points are important today, a few are particularly so.

1 to 8 channels of multiplexed audio:

The DTS system allows from one to eight channels of compressed audio in a single data stream. The bit allocation for each channel may be fixed or variable, depending on the demands of the application and the complexity of the encoder. There are many pre-defined channel configurations.

Custom channel configurations are possible. The total compressed and multiplexed audio data rate can vary from 32 kbps (kilobits per second) up to 4.096 Mbps (megabits per second), depending on a number of application-defined parameters. These might include the number of audio channels coded, the complexity of the encoding process, the sampling rate of the source PCM digital audio, and the data buffer size at the decoder.

This wide data bandwidth operating range allows DTS to find widespread use in many audio and audio/video applications ranging from telephone-grade voice audio at low data rates to multi-channel music formats operating at very high sampling rates for extended audio precision. That's impressive flexibility.

Sampling rate and audio word length:

DTS also allows sampling rates to vary from 8 kHz per channel up to 192 kHz per channel. As you remember, this results in a high frequency limit from 3.5 kHz to 90 kHz. The DTS encoding algorithm operates with 40-bit precision on 24-bit audio words. Using computer-generated test signals, and a 32-bit "floating point" decoder, the DTS algorithm can produce a dynamic range of up to 138 dB!

The "high density" sampling rates and word lengths are intended for professional and high-end audio applications. DTS provides a means of reproducing this higher quality audio in the home on digital media such as CDs and DVDs. .

Future-proof decoder:

Like Johnson and Pflaumer of HDCD fame, the DTS folks realized that a constant stream of "newer, better" decoders weren't in the consumer's real interest. After all, how many times would a customer step up to a "next generation" device before getting totally turned off?!?! So DTS decided on a strategy that allowed constant improvements at the *encoder* end but needed no hardware upgrades for playback *decoding*.

HOW IT WORKS

Like most digital processes, DTS encoding is a fairly complicated process. Unlike Dolby Digital, a lossy compression system which depends on proper application of psychoacoustic principles to present quality audio at low bit rates, DTS is currently used at higher bit rates. This means that two techniques useful for low-bit encoding, perceptual masking models and joint coding, are not necessary and therefore not used.

As we've explained elsewhere, perceptual masking refers to a phenomenon of human hearing where a loud tone tends to mask softer tones close to it in frequency. An encoding system like Dolby Digital uses these models to determine which audio information can be safely discarded prior to encoding.

Joint coding simply combines each channel's high-frequency content for greater coding efficiency. It's predicated on the fact that there is comparatively little high- frequency information (i.e., frequencies in the uppermost audible octave – above 10kHz) in more sources to begin with so combining them entails little if any sonic penalty.

DTS provides for future use of these perceptual techniques if lower bit rates are required.

DTS' key processes include sub-band filtering, linear-predictive coding (also known as "adaptive differential coding), vector quantization, and dynamic bit allocation. As if those esoteric topics weren't enough, there are additional factors like variable length coding and "encoder bit-rate iteration" to further confuse the issue.

Let's see if we can make these rather esoteric topics a bit more understandable.

We need to begin by noting that conventional digital audio devices, CD and DAT players, for example, use a linear PCM (Pulse Code Modulation) data structure for storing music signals. This data is recorded or stored in a simple, rigid format. The sampling rate and sample size (bit structure) are both fixed and the data is read from the digital media at a fixed bit rate.

	Sampling rate	**Sample size**	**Audio data bit rate [kbit/s/channel]***
CD	44.1 kHz	16 bits	705.6
DAT – Digital Audio Tape	48.0 kHz	16 bits	768.0
DVD-A (most common format)	96.0 kHz	24 bits	2304.0

* kilobits per second per channel

This simple coding strategy was very successful. CD audio, for example, has become the current consumer audio quality benchmark. DTS, however, believes that linear PCM coding is less than ideal for music or other sources that would benefit from a higher definition standard.

First, PCM, according to DTS, is not optimized for the characteristics of typical audio signals. Second, it is poorly matched to the sensitivities of the human hearing system. In DTS' opinion, these inefficiencies limit the audio quality of linear PCM-based systems in that they require comparatively huge amounts of data to represent high-quality audio formats, particularly DVD-A.

DTS points to the fact that linear PCM is ideal for coding full-scale, spectrally flat signals, such as white noise. However, since most music is not full-scale, and is spectrally tilted from low to high frequencies, PCM is a somewhat inefficient method for coding music. The diagram immediately above shows the spectral difference between white noise and music signals.

By substituting linear PCM with a more efficient coding methodology, DTS claims to be able to significantly improve the quality of digital audio with far less data than required by PCM. The following chart shows some of the differences.

	Sample rate	Sample size	Audio data bit rate kbit/s/channel
CD	44.1 kHz	16 bits	705.6
DTS	192 kHz	24 bits	705.6

DTS encoding begins by dividing the audible spectrum (up to 24 kHz) into 32 sub-bands. After this, each sub-band's output can be coded using LDC, a differential form of coding that predicts the next sample value.

A lot of DTS' vaunted "coding efficiency" comes from transmitting only a small residual error signal while occasionally updating the predictive filter coefficients. This depends on a process called "vector quantization," which groups various encoding strategies by how likely they might be used. In fact, the encoder often "parallel processes" a section of material and then selects the best of several independently computed strategies. This sometimes involves changing the form of the sub-band filters to best match the nature of the transient signals then being passed.

This information is then coded and bits assigned as needed. Accuracy depends on the number of available data stream bits, the subjective psychoacoustic models chosen, and "hiding" unwanted distortion products where they will be least intrusive.

In order to create a simple, future-proof decoder, DTS relies on two design features.

First, all the 'intelligence' of the system lies in the encoding stage – much like HDCD, you'll no doubt note. Thus, creating a consumer decoder is relatively simple. The decoder merely follows instructions within the audio bit stream generated by the encoder. This ensures that the encoding algorithm can be continually modified and improved, and that these improvements automatically benefit every consumer decoder.

Secondly, the DTS strategy includes a provision that allows additional audio data than was required at first. This was accomplished through the use of a sophisticated "syntax structure" that provides for future improvements or outright changes in an audio format. You'll see how this provision has come in handy when we discuss DTS ES and ES Discrete a bit further on.

DTS ON DVD

In the fall of 1999, DTS modified its original format at the request of many content providers who were concerned that the DVD standards, which mandated multi-channel Dolby Digital and two-channel PCM audio tracks, didn't leave enough room for DTS' original higher bit data stream. The reason was simple: More and more DVDs were being filled with "extra features" (interviews, "the making of . . ." specials, and the like).

> **Note*: The DVD-V audio standards, then being developed by the WG-4 working group within the DVD Forum, evolved in a charged political atmosphere. As a result of much lobbying, the DTS system was made an "optional" rather than a "required" standard. DTS, obviously, was very disappointed with this decision as it had little to do with technical merits as such.***

To accommodate its customers, DTS responded with a new version of its system that incorporated a data rate of 754 kilobits/second as opposed to the original's 1.5 *mega*bits/second. This meant an increase in the nominal compression ratio to approximately 8:1. It also fit into the ever-decreasing amount of disc real estate left over for audio after all the video data was encoded.

Some DTS partisans were disappointed with this development, a few going so far as to call it "DTS Light." But the fact is that the new format preserves most if not all of the original's sonic virtues and we see no reason whatsoever to criticize it.

You should note that, despite this new version, many DVDs use the original 1.5 megabit format. In fact, an increasing number of multi-disc DVD releases ("director's cuts," etc.) have more than enough room to accommodate the higher rate data. There's no hard-and-fast pattern here though. Each project's director or sound engineer makes a decision on a case-by-case basis.

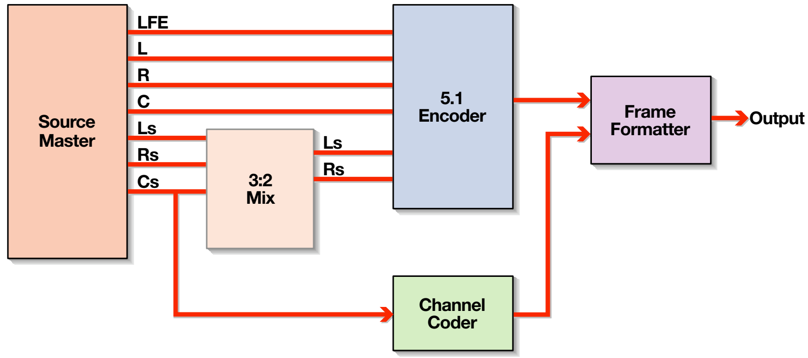
NEW DTS FORMATS

DTS ES and DTS ES Discrete

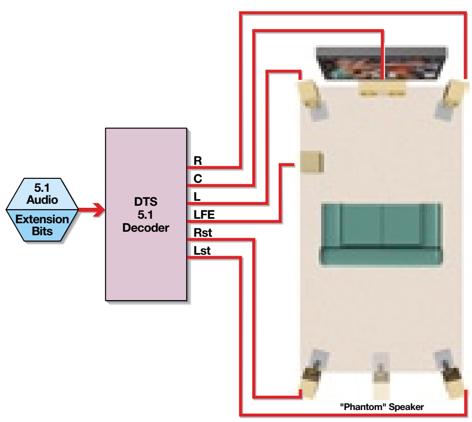
DTS ES (Extended Surround) is functionally identical to Dolby Surround EX in that it adds an additional "center surround" channel to the usual 5.1 format. The "center surround" channel is, just as in Dolby Surround EX, matrixed onto the two surround channels.

This "center surround" channel adds specificity to the surround soundstage when a movie director or sound engineer wants a more defined location than is possible with a normal 5.1 system. For example, a DTS ES-encoded soundtrack could place an explosion directly behind the audience rather than presenting a more amorphous "somewhere back there" localization.

Here are some of the technical details:



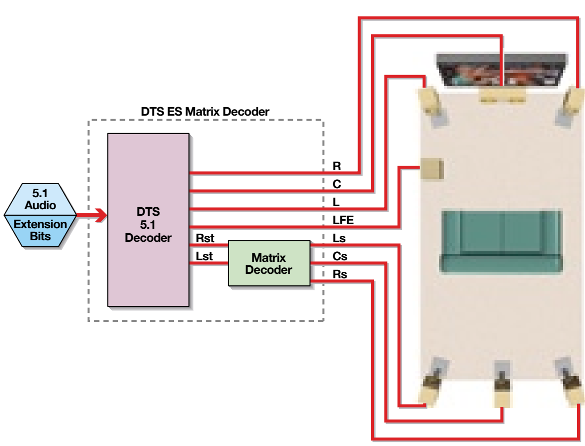
As you can see, a "mix encoder" mixes Ls, Rs, and Cs (Left Surround, Right Surround, and Center Surround) into Lst and Rst, which in turn are sent to the encoder. A channel encoder encodes the Cs (center surround) channel into extension data. Something called a frame formatter then appends the extension bits to the core data a frame at a time, creating a bitstream that carries 6.1 audio while maintaining backward compatibility with 5.1 decoders. The enhanced bitstream can then be recorded on a medium such as DVD.



Block diagram of a DTS decoder

A conventional 5.1 decoder decodes the ES bitstream a frame at a time by detecting the sync bit, reading the 5.1 formatted bits and ignoring the extension bits. The left, center, right, and LFE discrete channels are directed to the decoder outputs. The surround-total channels (Lst and Rst) which carry the three-channel mix, are directed to outputs Ls and Rs. This creates a phantom center surround signal that is heard between the Ls and Rs speakers. Thus users with 5.1 decoders will hear all the Cs sound elements in the mix.

Going one step further, a matrix ES decoder does the following:



Block diagram of a DTS ES decoder

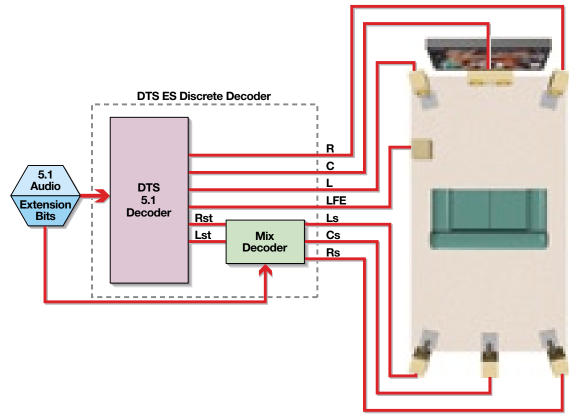
The basic playback configuration depicted above is enhanced by the addition of a matrix decoder and a center surround speaker. The matrix decoder decodes the surround-total channels Lst and Rst into three channels Ls, Cs, and Rs. One drawback? The dematrixed audio signals exhibit some crosstalk and phase distortion that are, unfortunately, typical of matrix decoding. There is no free lunch.

DTS ES Discrete carries this one step further. Using DTS' additional bandwidth to greatest advantage, an ES Discrete soundtrack contains a fully independent "center surround" channel that is encoded as a totally separate entity without "piggybacking" on the two main surround channels. The advantage here is theoretically better placement of rear channel effects because there is no crosstalk or phasing errors. Plus, of course, you get fully discrete 6.1 playback (with properly encoded software, an ES Discrete decoder and extra speakers, of course!).

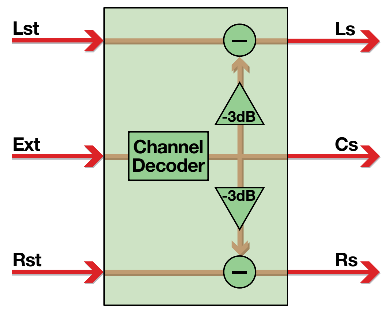
Here's how it works. A 5.1 decoder reads the 5.1 audio from the bitstream. It ignores the "extension bits" buried in the datastream while it decodes the L, C, R and LFE signals, which are then passed on to the decoder outputs. Lst and Rst (Left Surround Total and Right Surround Total) signals are passed to the mix decoder which reads the extension bits. The mix decoder then decodes the Cs signal and uses it to separate surround signals Ls and Rs from Lst and Rst. (Most decoder ICs today perform this as a single process.)

The mix decoder also weights Cs (reduces it by 3 dB), then subtracts it from the Lst and Rst signals to remove all traces of the center surround channel. This leaves only the discrete Ls and Rs signals.

Perhaps surprisingly, the circuit is easily expandable to accommodate more than three surround channels by using additional channel decoders, multipliers and summing nodes.



Block diagram – DTS ES Discrete decoder



The ES Discrete "Mix Decoder"

DTS 24/96

As we've explained elsewhere in this book, extended-bit digital formats (primarily those with 20- and 24-bit words at sampling frequencies equal to or higher than 48 kHz) grew out of the limitations of "Red Book" audio, the CD standard.

As explained elsewhere in this book, CDs preserve digital data in the form of 44,100 16-bit words per channel every second. This means that both ultimate resolution and the ability to preserve ultra-high frequency components of a source are somewhat compromised.

Audio professionals have long enjoyed the aural benefits of much higher bit densities, with 24-bit words as a rate of 96,000 per channel per second being today's de facto standard. There are several theoretically audible advantages arising from these high data rates. On the undisputed side, we can definitely point to the fact that greater bit depths provide extended dynamic range and that higher sampling rates allow wider frequency response. The use of anti-alias and reconstruction filters with more favorable aural characteristics results in more pleasing sound.

The advent of DVD meant, among other things of course, that it became possible to deliver 24-bit, 96 kHz audio into the home. However, this data rate was restricted to two-channel-only sources: multi-channel audio was necessarily compressed via Dolby Digital or DTS to leave enough room on the disc for video information.

In addition to suffering from the basic inefficiencies of PCM encoding, DVD-Audio did not totally solve the problem as

1) You had to buy a new DVD-A player
2) You couldn't get high-density digital audio from the digital output but were

restricted to the analog signal provided by whatever D/As a manufacturer included in the player.

DTS 96/24 offers the following:

1) Sound quality equal to the original 96/24 master
2) Full backward compatibility with all existing decoders. (Existing decoders will produce only a 48 kHz signal.)
3) No new player required: DTS 96/24 can be carried on a DVD-Video disc or in the video zone of a DVD-Audio disc and thus be accessible to all DVD players.
4) 96/24 5.1 channel sound with full-quality full-motion video, for music programs and motion picture soundtracks on DVD-video.
5) 96/24 sound quality through the S/PDIF output available on almost all DVD players to allow the use of external, very high quality D/A converters and associated analog electronics chosen by the users.

DTS Neo:6

Neo:6 brings many of the sonic advantages of discrete multi-channel sound to the world of matrixed two-channel soundtracks and music recordings: the large library of matrix surround motion pictures available on disc and tape and to analog television broadcasts. In that respect, it is similar to Dolby Pro Logic II but Neo:6 did come to the party first.

Typical matrix decoders derive a center channel and a mono surround channel from two-channel matrix stereo material. Although better than a simple matrix in that it includes steering logic to improve separation, its monaural, bandwidth-limited surround is disappointing to users accustomed to discrete multi-channel.

For those interested in better sound from older sources, Neo:6 offers several important improvements.

1) **Stereo matrix to 6.1 or 5.1.** Neo:6 provides up to six full-bandwidth channels of matrix decoding from stereo matrix material. Those with 6.1 and 5.1 systems will derive six and five separate channels respectively, corresponding to the standard home-theater speaker layouts. (The ".1" LFE or subwoofer channel is generated by bass management in the preamp/processor or receiver.)
2) **Extended Surround**. Neo:6 provides optimum decoding of Extended Surround matrix soundtracks and can also generate a back channel from 5.1 material.
3) **Natural sound field** with precise localization. Neo:6 technology allows various sound elements within a channel or channels to be steered separately, and in a way which follows naturally from the original presentation.
4) **Music mode**. Neo:6 offers a music mode to expand stereo non-matrix recording into the five- or six-channel layout in a way which does not diminish the subtlety and integrity of the original stereo recording.

COMPATIBILITY NOTES

Today, DTS-encoded audio tracks are available on both CDs and DVDs. (There are a few DTS-encoded laserdiscs still around but you'll have to look very carefully to find them – as you'd have to look carefully to find *any* laserdiscs!) Strictly speaking, these DTS-encoded "CDs" really aren't CDs as their data format is at odds with the standards set by the Sony/Philips "Red Book" that defines the CD: They don't follow the Red Book format. And, of course, you'll need a DTS decoder to listen to them.

CAUTION: If you accidentally place a DTS-encoded "CD" into a conventional CD player or CD transport/DAC combination and listen through the analog outputs, you may well damage your loudspeakers. That's because, with no intervening DTS decoder, conventional digital-to-analog converters interpret the DTS-encoded data as full-scale white noise. If your system's volume control is set anywhere near normal playback levels, you'll hear something like a berserk version of Niagara Falls for a brief moment before your speakers' voice coils burn up. A later – and far more predominant – version of the DTS-encoding algorithm reduced this white noise level by 12 dB. Although that change also reduced the "destruction potential" significantly, you'll still jump if your playback system lacks the appropriate DTS decoder.

> ***Some sonic history**: Some persnickety audiophiles denigrated some early multi-channel music mixes – even if they were truly stunning in overall sound quality – for their occasionally over-the-top production values. Level shifts – usually up – and zippy EQ choices seemed to be the then-current rage. Whatever your sonic sensibilities may tell you, remember that EQ and rear-channel levels are choices made by producers and recording or mastering engineers. They are not – as they have been presented by some – artifacts of a particular coding system.*

The situation with DVD players is as follows: All current – and almost all existing – players can pass a DTS signal through the digital output to a decoding receiver or preamp/processor. Some contain DTS decoders as well. Newer models almost always have DTS decoding capability built in. Depending on which model you've chosen, you might have to enter the player's menu system to enable DTS playback. (This might need to be done only once but you should check your owner's manual for details.)

Some of the earliest DVD players will *not* output a DTS datastream. That leaves you with the option of listening to the two-channel PCM tracks through the player's analog outputs or the Dolby Digital track.

PREVIEW

Exactly what is THX? If that seems like a trick question, it isn't. THX has gone through a number of evolutions since it first appeared as a set of specifications in 1988 and, finally, in a product in 1991.

We'll outline some of the background and trace the technology from the original THX spec through THX Ultra and THX Select, look at THX Surround EX, and then at the new Ultra2 spec.

By the end of this section, you'll know it all!

Some people seem to think that THX is some sort of "super surround," that it's better than either Dolby or DTS in some (deeply mysterious?) way.

Nothing could be further from the truth.

In reality, THX means many things: a set of standards and an inspection program designed to ensure top-quality sound at THX-certified theaters; a quality control program for film prints; a film-to-video transfer program to ensure the best picture and sound quality in DVDs; and, most importantly at least for our purposes here, a way to make sure that home theater systems sound as close as possible to the dubbing stage on which the original film soundtrack was mixed.

Nothing more. Nothing less, either. When you think about it, that's a pretty tall order.

Notice that we said nothing about multi-channel sound encoding or decoding. Or getting the correct signal to the proper speaker of your home theater array. Dolby Digital, DTS, and other "pipeline" technologies do just fine in that department, thank you very much.

Instead, THX builds on these important efforts with a series of technologies, some electronic and some acoustic in nature, that make it easier for us to hear at home what the director and sound engineer intended us to hear in a real theater.

In short, THX attempts to bridge the audible gap between the Bijou and your basement.

WHAT DOES THX MEAN, ANYWAY?

The term THX has two possible origins. Most people think it stands for "Tomlinson Holman's Experiment," a thoroughly reasonable view as Holman directed the research that resulted in many THX standards. (See the paragraph below for a bit more information.) Film buffs, on the other hand, remember that George Lucas' first commercial effort was the 1971 sci-fi flick, *THX 1138*. For a time, THX was quite vociferous in stating that the film inspired the name, particularly after Holman left Lucasfilm to pursue other interests. But following THX's recent spin-off from Lucasfilm, Holman is once again getting credit. Believe whichever story you want to – it doesn't make a lot of difference in the real world. We just thought you'd like to know.

A BIT OF HISTORY

The whole THX project began in 1980 when George Lucas hired Tomlinson Holman, an audio engineer of some note (remember the Apt/Holman preamp?) and then professor of film sound at the University of Southern California. The *Star Wars* whiz kid wanted Holman to research every influence on film sound including mixing practices and movie theater acoustics. The goal was to find practical ways to make movie theaters sound more like dubbing stages and, incidentally, more like each other.

The Problem

Most of this research involved the professional side of film sound and resulted in several programs to ensure high sound quality in a theater. However, the THX team soon discovered that no home environment allowed them to hear a film's soundtrack as they routinely heard it on a dubbing stage. (The dubbing stage, a cross between a movie theater and a recording studio, is where all a film's diverse sonic components are balanced and mixed with each other to make up the final soundtrack.)

This discrepancy was of major interest as dubbing stages worldwide all sound remarkably similar thanks to our old friend, SMPTE 202M. This isn't surprising as a similar acoustic environment is mandatory in an industry that routinely shuttles soundtracks around the globe for various post-production efforts.

As you can easily imagine, it would be sonically disastrous for one mixing stage to have a radically different acoustic signature from another: The sound engineers would then have no way to avoid sonic discontinuities as they added their touches to the final mix.

The problem, as gradually defined by the THX staff, centered around the fact that no two home environments sounded the same and none came close to the acoustic environment of a standard dubbing stage. By 1987, Holman's team had developed several approaches and had built prototypes to test their validity. The first THX specs were formalized in 1988 and products from several consumer electronics manufacturers appeared in 1991.

Specifically, Holman's researchers identified the following troublesome areas:

1) Low "intelligibility": The dialog was "muffled" and there was a general loss of details

2) Blurred frontal image: You couldn't tell exactly where sounds were coming from.
3) Very "bright" sound
4) Very constrained surround soundfield: It never seemed to "escape" the surround loudspeakers.
5) Uneven sounding pans caused by the different way we hear sounds originating in front of us compared to how we hear sounds coming from the rear
6) Poor dynamic range performance: Loud sounds weren't very loud and soft sounds sometimes got "buried" in noise.

The Solution: "Original THX"

At The Processor . . .

The THX team's investigations centered on ways to correct these discrepancies and resulted in the following **circuitry developments** that were **required of all THX processors**:

1) THX Re-Equalization™ corrects for the overly bright quality most film soundtracks display when played in a home environment.

Re-Equalization is just what its name says. Re-EQ uses a "shelf" filter to cut high frequencies so film soundtracks don't sound so harsh and "zippy" when you play DVDs at home. Re-EQ is really useful because most movie soundtracks contain extra high frequency energy so they'll sound good in a large theater where the audience is much further away from the speakers than they would be in a home environment.

2) THX Timbre Matching™ smoothes transitions between front and surround speakers

Timbre Matching (pronounced "**tam'-**bor" by the way) evolved from the fact that our pinnae (the outer channeled portions of our ears) filter sounds coming from behind us far more radically than they do to sounds coming from in front of us.

Why is this important for a home theater system? Simple. Suppose you have a soundtrack with a "flyover," a scene in which a plane, spacecraft, helicopter, whatever, moves from the front of the room to the back (or vice versa). Without Timbre-Matching, for example, we *might* be conscious of a change in tonality as the sound effect panned to the rear. (Notice that "*might*" is italicized. There's a reason for this.)

Timbre-Matching minimizes those subtle sonic differences so they're not as jarring or distracting to us. Technically, the Timbre-Matching circuit is a "multi-pole filter" with a rather complex curve (some frequencies attenuated more than others). Interestingly enough, Timbre-Matching's effects are more noticeable with monopole surround speakers than with dipoles. (You'll find more information on dipoles further on in this Section.)

3) **THX Decorrelation™** enhances the apparent spaciousness of Dolby Surround's mono rear channel signal.

   Decorrelation was originally developed to enhance the apparent spaciousness of mono surround information – exactly what you got with a Dolby Surround-encoded soundtrack decoded by a Dolby Surround or Pro Logic decoder. Even when we played this mono surround information through two speakers, it often sounded very constrained and non-enveloping.

   Decorrelation uses the two separate surround amps (and speakers) very creatively by splitting the mono surround signal and then subtly shifting phase and amplitude in each derived channel just enough to create a much broader sense of "surround spaciousness." This was a big improvement for those mono surround tracks.

   Of course, Dolby Digital and DTS (as well as any other discrete multi-channel format) can have real stereo surround information already. This would make Decorrelation unnecessary. So THX modified the circuit. Now it goes by the name **Dynamic Decorrelation**™ but the only difference between it and the older version is that Dynamic Decorrelation monitors both surround channels continuously. When there's "differential information" (engineeringese for different information in Left Surround and Right Surround channels), Dynamic Decorrelation stays out of the picture. It does nothing. But when it detects mono surround information (exactly the same signal in both surround channels), it swings into action to create the "just different enough" effect we need for more enveloping ambience.

   How useful is Decorrelation – Dynamic or otherwise? That's sort of difficult to answer. For mono surround tracks, this technique can add a welcome sense of spaciousness, particularly with ambient effects like rain falling in a jungle, etc. With stereo surround, as we've already explained, Decorrelation is moot anyway.

4) **THX Bass Management** improves the dynamic range of the entire system by making sure that each system, regardless of what speakers a consumer uses, delivers the most accurate and least distorted bass energy possible.

Bass Management is a complex topic but you do need to know something about it. Here's the short version.

Bass frequencies tax many so-called "full range" loudspeakers. The audible effects of sending more bass to a speaker than it can comfortably handle (a form of "overload") include distortion and reduced dynamic capability. To avoid them, THX processors include a circuit called a crossover that divides the frequency range and routes bass frequencies to the subwoofer and directs the rest of the audible range to the wide range speakers which then have to reproduce mid and high frequencies only.

Before the crossover can do its job, however, you need to tell it what your speakers are capable of. Are they "true full-range speakers" that have real bass capability? Are they small "satellite" speakers with comparably small woofers that lack deep bass capability? What about your center channel speaker? And surrounds?

The THX bass management system needs answers to all these questions before it can decide how to best route the bass information. Some setup screens get this information indirectly by asking if your speakers are "large" or "small" as if size alone determined bass capability. But if you substitute "relatively unlimited bass capability" for "large" and "limited bass capability" for "small," you'll be on the right track. Once you've answered these initial setup questions, Bass Management takes over and directs low frequencies according to the table below.

Here are a few hints to help you answer the questions you'll be asked:

1) First, find out if your main Left and Right speakers are full range designs (with good bass capability) or have limited low bass output. Use your processor's setup menu and select the proper choice.
2) Next, tell your processor whether your center channel speaker is full range or has limited bass capability.
3) Then, input the bass-handling capabilities of your surround speakers.
4) Finally, tell your processor whether or not your system includes a subwoofer.

Once you've done that, Bass Management takes over and routes low-frequency information as follows:

Main L/R	Center Channel	Surround Speakers	Subwoofer	Bass Information from . . . goes to . . .
Full range	Full range	Full range	On	LFE only . . . to Sub
Full range	Full range	Full range	Off	LFE . . . to Main L/R
Full range	Full range	Limited	On	LFE + Sur. . . . to Sub
Full range	Full range	Limited	Off	LFE + Sur. . . . to Main L/R
Full range	Limited	Full range	On	LFE + C . . . to Sub
Full range	Limited	Full range	Off	LFE + C . . . to Main L/R
Full range	Limited	Limited	On	LFE + C + Sur . . . to Sub
Full range	Limited	Limited	Off	LFE + C + Sur. . . . to Main L/R
Limited	Full range	Full range	On	LFE + Main L/R . . . to Sub
Limited	Full range	Limited	On	LFE + Main L/R + Sur. . . . to Sub
Limited	Limited	Full range	On	LFE + Main L/R + C . . . to Sub
Limited	Limited	Limited	On	LFE + Main L/R + C + S . . . to Sub
Limited	Any	Any	Off	No crossover function (Bass Management off)

Bass Management's other chief benefit lies in the fact that it allows consumers to choose smaller, more easily concealed speakers for LCR applications without suffering any loss of bass impact. Bass Management also recognizes the often-conflicting placement requirements for low bass speakers and higher range transducers.

5) **Miscellaneous**: Those technologies are THX's "big electronic guns." But, as you might expect from the comprehensive research performed by Tom Holman and his crew, other things are also involved. THX requires **time delays on *all* channels,** a more sophisticated – and more expensive – approach than Dolby's Time Alignment. The THX version is called Loudspeaker Time Synchronization^{tm}. There's also a **limiter circuit** for the bass called Bass Peak Management^{tm} to prevent overdriving loudspeakers.

Amplification

As you've no doubt guessed, THX wasn't satisfied with what processing alone can do to narrow the gap between dubbing stage and your home.

There are a number of amplifier specifications also. The main points here have to do with an amplifier's ability to deliver reliable long-term output sufficient to assure undistorted output under very demanding conditions. These specifications, while interesting, do not constitute anything new technologically and are well known

– if not rigorously followed – by any competent amplifier designer. Consequently, we won't – with one exception – delineate them here.

That exception is the *input sensitivity* spec. This spec simply identifies the strength of the input signal needed to drive the amplifier to full output. The reasoning behind this has to do with the fact that an amplifier is what we call a *constant gain* device. In other words, it will increase signal strength – both voltage and current – by a fixed amount only. (That's why some amplifier manufacturers will tell you that their Model X amp will increase gain, for example, by 28 dB.) Any variation in loudness is really a function of the source or the volume control setting of a processor or the preamp section of a receiver, etc.

And *that* brings us to the pesky – and rarely understood – "0 dB" reference mark on the volume control of every THX-certified processor or receiver.

We all know that "loud" is subjective. Just ask a person trying to sleep while the faucet in the bathroom keeps dripping. But "0 dB" isn't subjective at all *if* your entire system is THX-certified. If it is, then placing the volume control at that point will produce playback levels in your home that are the same as those used to master the soundtrack in the first place. Typically, an "0 dB" volume control setting produces average levels around 85 dB SPL. (Sound Pressure Level). This is about 20 dB louder than conversation levels you experience at home and only about 35 dB below the threshold of pain. That's why very few people with THX-certified systems ever listen at "0 dB" – unless they're trying to impress an innocent victim. Or annoy a neighbor.

Front LCR Loudspeakers

> ***Note: Newer THX speaker standards – Ultra2 and Select – are different in some ways. However, once you understand the original specs, you'll find that evolution more understandable.***

In addition to the electronic refinements we've just discussed, the original THX specs included the following considerations for the **front Left, Center, and Right Channel (LCR) speakers.**

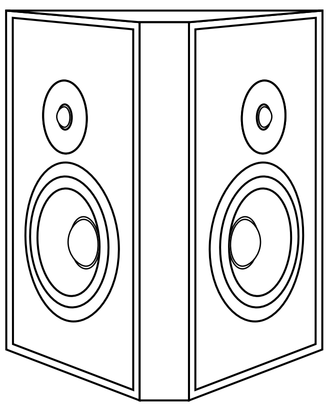
1) **Focused vertical directivity** to minimize floor and ceiling reflections. This means that not as much sound would "bounce" back to your ears. Excessive floor and ceiling reflections would contribute to confusion of the front sound field and possibly reduce dialog intelligibility. There are several ways to do this, the most common being to sandwich a tweeter between two midrange drivers in what is called a "D'Appolito configuration," named after Joseph D'Appolito, the acoustics designer who first described its benefits.
2) **Wide horizontal dispersion** for improved coverage of the listening area and a wider optimum viewing/listening area.
3) **Magnetic shielding** to allow wide placement options without affecting nearby

TV screens. Proper magnetic shielding protects TVs from the interference generated by loudspeaker magnets. This interference affects direct view TVs especially. Placing an unshielded speaker too close to a conventional set can permanently damage the picture tube. (Rear projection sets are less affected, front projectors almost not at all.)

Surround Loudspeakers

Originally, THX-certified surround speakers had to exhibit:

Dipole Surround Speaker



The rear set of drivers (woofer and tweeter) operates "out of phase" from the front set. When properly placed in a room, this produces a very diffused soundfield in which it is all but impossible to determine exactly where the surround effects are coming from.

1) **Dipolar design** to enhance the sense of spaciousness and better envelop listeners and thus involve them in the movie experience.

   Of course, understanding this begins with a clear view of what a dipolar speaker is.

   Dipolar design enhances the sense of spaciousness and is better as "enveloping" listeners in the movie experience. Why? A dipole's key feature is the midrange null between the two faces of the speaker. By aiming the null at the prime viewing/listening position, there is less chance that the audience will be distracted by easily-localizable sounds coming from the surround speakers, thus breaking their concentration on screen action.

   THX firmly believes that dipoles are the best way to mimic the effect of a theatre's surround speaker array in the home. And, as you might expect, dipolar speakers enhance the same spaciousness effect that Decorrelation attempts to create electronically.

   The use of dipolar surround speakers is hotly debated. Many respected acousticians and speaker designers oppose their use. Interestingly, THX recently genuflected to commercial opportunities when it allowed directional surround speakers as part of its more recent and more affordable "Select" certification program.

2) **Flat power response** to better match the sound from the front speakers. Here, flat power response simply means that the total energy a surround speaker radiates into a room is very smooth and even, even when measured off-axis. This means that when you measure a surround speaker's output at the viewing/listening position, it will probably be very similar to the on-axis measurements of the front LCR speakers. This assures smooth continuity from front to surround speakers as a sound pans from the screen towards the rear of the room or from the rear to the front. As you can imagine, flat power response couples with the electronically implemented Timbre Matching circuitry described above to enhance front-rear coherency.

3) **Side wall placement capability**. Although side wall placement wasn't an absolute THX requirement, it was highly recommended as it allowed the listener to fully benefit from a dipole's spaciousness while minimizing direct sound – and the ability to localize the ambient sounds – mainly produced by surround speakers.

The Subwoofer

Recognizing that low-frequency effects are an important part of the movie experience,

THX also developed stringent specifications for subwoofers:

1) **In-room frequency response** had to extend to 20 Hz. Note the words "in room." That's not the same thing as "anechoic" response so if you see a spec for a THX-certified subwoofer that extends to only 35 Hz or so, don't panic. The in-room response requirement takes something called "room gain" into consideration. Room gain lifts very low-frequency response compared to an anechoic measurement.
2) **Minimum impedance** specifications meant that THX-certified subwoofers wouldn't tax well-designed amplifiers (or the amplifier sections of "powered subs").
3) **Magnetic shielding** was required to insure placement flexibility.
4) **Extended power handling capabilities and minimum sensitivity specifications** assured sufficient output levels to properly reproduce bass dynamics.

Summarizing the original THX specs

As you can see, these specs made "home THX" a system-wide approach. What you should remember also is that these original specs were based on the most popular surround sound format of the day – Dolby Surround as decoded by Dolby Pro Logic circuits. The original THX spec couldn't apply to the entry-level Dolby Surround decoders (i.e., non-Pro Logic designs) then available because Dolby Surround decoding did not supply a center channel.

> **Note*****: THX specifications were soon developed for other components in a home theater system: DVD players, DVD discs, projection screens, interconnect cables, etc. Today, there are THX specs for multi-media speaker systems for computers and for automotive sound systems.***

REINFORCING AN IMPORTANT DIFFERENCE

As you can now see, THX and Dolby (or DTS, or Circle Surround, or Extended Surround) have little to do with each other. The reason they're often confused is that you can find Dolby Digital or DTS or … in all THX-certified processors.

Simply remember that a surround sound format (Dolby, etc.) and THX are *complementary*: One is *not* better than the other. They address *totally* different links in the technology chain that brings home theater excitement to you.

Briefly stated, Dolby, DTS, etc., address the following question: *How does my system know what sounds go where?*

THX, on the other hand, *deals primarily with the acoustical differences between a dubbing stage and the home*. THX operates only *after* Dolby or DTS processing (or whatever format) has already determined directionality. THX tries to make those sounds as close as possible to what the movie director or sound engineers intended.

"SELECT" VS. "ULTRA"

Prior to 1999, there was only "THX" and "not-THX." Things became a lot less simple that year as the original THX spec evolved to Ultra status and a new specification, THX Select (called "THX Light" by some cynics), emerged.

The original THX spec was technically challenging and expensive to meet. Consequently, products that earned the certification were relatively expensive and comparatively few of them made it home from dealer showrooms.

As a licensing organization, THX saw additional opportunities in a less costly – or Select – certification program.

Originally, THX (now dubbed "Ultra") was developed for a 3,000 ft^3 room (approximately 22′ x 17′ x 8′). And those well-to-do customers who bought the first THX products probably lived in spaces that echoed those generous dimensions. But urban apartment dwellers – or those who simply lived in less palatial dwellings – were a different story. In fact, the performance levels mandated by the original "large room" spec were truly excessive. After all, the job of moving such a large mass of air demanded substantial speaker output capability and amplifier power.

THX Select specs were developed for a "downsized" room (hence the "Lite" designation). The ideal room size for a Select system was approximately 2,000 ft^3 in volume (approximately 13′ x 19′ x 8′). The smaller air mass reduces the demands on loudspeakers and amplifiers. In fact, most of the differences between THX Ultra and THX Select dealt with only speakers and amps. Processor requirements were essentially identical – they both shared all the technologies we've already outlined: Re-Equalization, Timbre-Matching, Dynamic Decorrelation, and Bass Management, etc. True, Ultra specs are a bit tighter in some areas but it is unlikely that even a relatively critical listening session would reveal something essential to most listeners (THX

Surround EX – something we'll talk about shortly – a possible exception).

System Differences: Ultra vs. Select

An Ultra-certified system must produce continuous output of 105 dB SPL in a 3,000 ft^3 room for three hours while a Select amplifier/speaker combination must produce peak levels of 105 dB in a 2,000 ft^3 room (no time duration specified).

These are substantial differences.

Ultra-certified systems play *louder*. A continuous output rating is much more demanding than an instantaneous peak rating. Ultra-certified systems play *longer*, too. The Ultra spec mandates a minimum of 3 hours without stop. Select certification doesn't include any long-term output tests at all. And don't forget that bigger "Ultra" room size either! Put an Ultra system and a Select system in that larger room and play a soundtrack at a substantial volume setting. You'll definitely hear a difference in system dynamics. However, if you're playing back at levels 10 to 20 dB below that (in)famous "dB" reference mark, you might be hard-pressed to hear any important differences.

Remember that Ultra-certified front loudspeakers have narrow horizontal dispersion. Select specs are much looser in this regard. But remember that there are other areas of speaker performance considerably more important than directivity. Interestingly, frequency response parameters for Ultra and Select are the same. In a smaller room, the listener will be sitting closer to the speakers so the Select spec may be sufficient.

Incidentally, don't be too concerned about reflections from the side walls. Yes, they do exist. Yes, they can influence intelligibility. But, yes, they are much easier to control than floor and ceiling reflections. Properly placed bookcases, acoustical panels, or drapes that minimize or break up reflections are only three of many solutions.

Surround Speaker Specifics

THX Ultra specs permit only dipolar surround speakers.

THX Select specs allow more conventional monopole (or directional) designs for surround use. Obviously less expensive, this option allows a manufacturer to design a more cost-effective surround speaker. Although many prefer dipoles for movie surround reproduction, the *music industry* has opted strongly for directional surround speakers.

In any case, proper speaker placement, orientation, and *system balancing* are far more important than the type of speaker you use.

Note: ***Both Select and Ultra allow "In-Wall" and "On-Wall" front speakers. If you have an aesthetic objection to free-standing speakers, remember this.***

THX SURROUND EX

THX Surround EX, jointly developed by Dolby Laboratories and Lucasfilm, is a part of the Ultra spec. To an extent, it is a surround sound format in that Surround EX includes a "Surround Back" channel that can be decoded and played by a Surround EX capable system.

Surround EX is not, despite what you might read elsewhere, a "6.1" format. The surround back information is NOT discrete but is matrixed onto the two discrete surround channels of a true "5.1" source. For this reason, some have called Surround EX "Dolby Pro Logic for the rear channels." Thus, Surround EX is, at best, a "5.1+" format that can, when the additional speakers are properly placed, improve our ability to localize sounds coming from behind us.

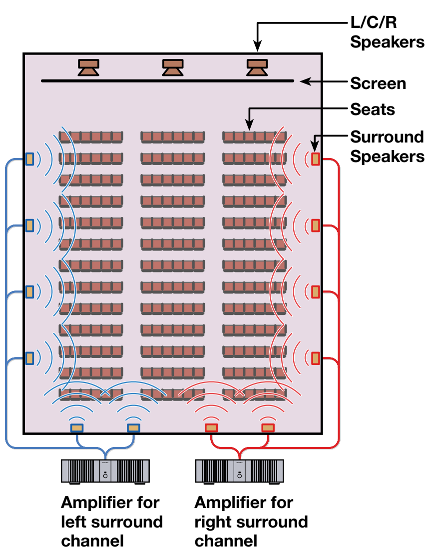
Surround EX originated in the film community. One of the key players was Gary Rydstrom, the award-winning sound wizard at Skywalker Ranch, the technical center for Lucasfilm and, at the time, THX. Gary wanted to offer a movie audience the sense that something was directly behind them, rather than just "back there somewhere." Even Dolby Digital's discrete surround channels didn't have that capability, particularly for latecomers who didn't get those prized center seats at the local Bijou.

The Skywalker crew approached Dolby Laboratories for help. The answer, they both knew, couldn't just be new technology. Compatibility was a major issue, too. In other words, whatever they came up with had to be "backwards compatible" – had to work with older equipment as well as in the newest, most up-to-date theaters in the world.

Surround EX at the Movies

To understand how ingenious – and economical – their answer was, let's look at a typical theatrical sound system.

Surround EX uses the same theatrical loudspeaker array in a different manner. Rather than divide the 12-speaker surround array into just two groups, Surround EX divides the array into three. Speakers along the left wall reproduce only "left surround" data, speakers on the right wall reproduce only "right surround" data, and the speakers along the back wall are reserved for "surround back" information carried in a special Surround EX-encoded soundtrack on the film.

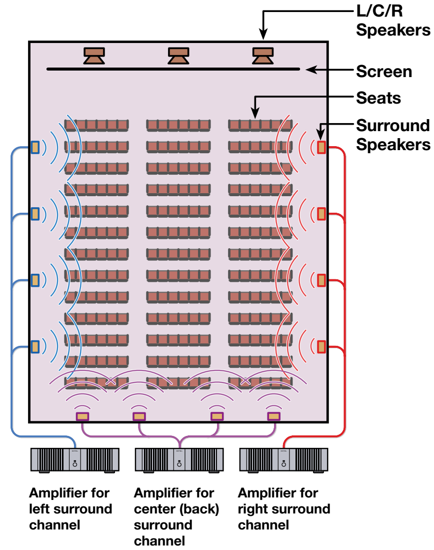


In a theater, surround information comes at us from many directions – it's hard to localize. That's good if the film's sound crew is using the surround channels to create a sense of ambience. It's not so good if the director or sound engineer wants us to "lock on" to a specific point where a particular sound appears to come from.

Here's how the same theatrical speaker array is arranged for Surround EX.

Notice that Surround EX in a theater doesn't require any additional speakers. It just uses the existing speakers in different groupings. One of the trick aspects of a Surround EX theater installation is that these speaker groups can be instantly reconfigured to reproduce a standard Dolby Digital or a Surround EX-encoded film! (In theory, a single soundtrack could contain both standard Dolby Digital and EX-encoded portions and the speaker groups – and amplification, of course – would track these changes as the film progressed!)

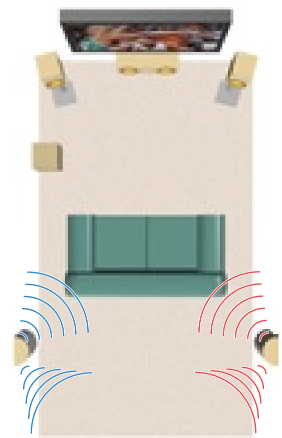
From a historical perspective, we first heard Surround EX on May 19, 1999, when *Star Wars: Episode One – The Phantom Menace* made its theatrical debut.



Surround EX gives the film's creators a "surround back" channel so the audience can quickly identify a sound coming from directly behind them.

Surround EX Comes Home

To add Surround EX capabilities to a standard THX system, we must add two speakers – and the amplifier channels to drive them, of course – on the rear wall. And we'll need special Surround EX-encoded software to fully benefit from the new technology. Initially, Surround EX software was scarce. At first, Skywalker Ranch was the only facility with Surround EX encoding capability and their maximum capacity was 13 to 15 soundtracks a year. Now, other studios have this encoding capability and EX-encoded films are more common today.



Here's a typical THX speaker installation. Notice how the surround dipoles are oriented.

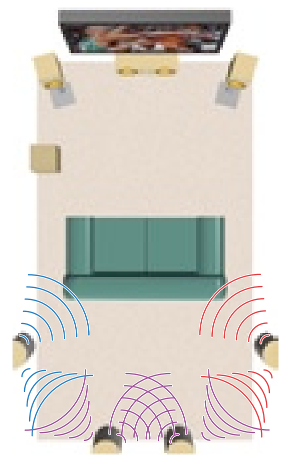
A Technical Puzzle

The initial shortage of EX-encoded software and the hardware platform required for EX together provide an interesting glimpse of home theater's future.

EX requires 8 separate signal paths: Three across the front (LCR), two conventional surround channels (LS and RS), two back channels (LCS or Left Center surround, and RCS or Right Center surround), and the LFE output. (Yes, the EX spec mandates *two back* outputs even though the signal sent to each may well be mono!)

Many D/A converter ICs today are stereo designs. So a manufacturer will need 4 of them. When you add the microprocessors that process Dolby Digital, DTS, and THX to the additional amplifiers needed for the Surround EX "back" channels, you're left with an interesting question: Should this complex and highly capable hardware platform come into its own only for Surround EX sources?

The answer to that is an emphatic "No."



Notice how the polarity markings on the dipoles complement each other. This arrangement gives the best spatial presentation for Surround EX-encoded material.

Surround EX could drive home theater to full 7.1 capability. That's because it's easy to use the number-crunching power of today's microprocessors to create "back" information for a non-EX- encoded source. And that opens the door to make use of the existing amps, D/A converters, and back speakers on a full-time basis.

A Non-Dipole Musical Future?

As we've already mentioned, very few music recording studios use dipolar speakers for "surround" applications even though interest in multi-channel music is expanding at an almost-geometric rate. Instead, engineers and producers are using the same full-range directional speakers in back as they use in front. And they feel very strongly that music should be heard through speakers of similar dispersion characteristics.

Surround EX may answer that dilemma very nicely by giving users the choice between dipolar surround speakers for film, directional surrounds (mounted on the rear wall) for music, or the full speaker complement for EX-encoded software!

ENTER ULTRA2

The final development we'll deal with here is the introduction of the Ultra2 specification. Ultra2 appeared in 2001 and replaced Ultra as the "high end" of THX's spectrum of specs. (The appearance of Ultra2 did not affect the Select spec at all – it remains current.)

Ultra2 addresses the growing importance of multi-channel *music* in addition to providing accurate reproduction of film soundtracks. In today's home theater market, particularly considering the increasing presence of SACD and DVD-A, the ability to handle both music and movies *transparently* (without much intervention from the system's owner) is more and more important.

Ultra2 design objectives include

- Excellent performance with both movies and music. Ultra2 includes automatic detection of source type and selection of processing strategies. (Manual override possible, too.)
- The use of a single loudspeaker array for both movies and music
- To provide a wider listening area for optimum sound
- Decreased dependence on a room's acoustic properties

Perhaps surprisingly, Ultra2 differs from Ultra in fewer ways than you might expect. But these differences are important ones. Ultra2 specs cover processing, amplification, and loudspeakers. We'll outline the important differences so you'll see them easily.

1) For "screen speakers" (L, C, and R), the requirement for very tightly controlled vertical (up and down) dispersion we saw in the original THX and THX's Ultra specifications has been considerably relaxed. Lest some see this as a backing-off from tough but necessary standards, THX has simultaneously tightened the requirements for mid-range response and octave-to-octave balance.

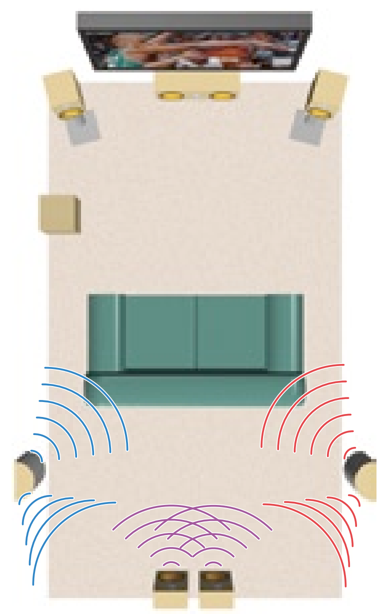
In addition, the requirements for smooth off-axis response are now more stringent. This combination insures that the speaker is more accurate generally and that the speaker's timbre does not change appreciably as you move off to the side.

2) The specification for subwoofer response has been tightened. The Ultra2 spec now calls for response down to 20 Hz with only a 3 dB drop-off. This is tighter than the original spec which called for anechoic response down to only 35 Hz. In addition to providing almost another octave of carefully controlled bass output capability, another benefit of this spec is that the acoustical "match" between the subwoofer and the side range "screen speakers" is more accurate.

3) Monopole surround speakers are now permitted. This is in sharp contrast to the original THX and the THX Ultra spec. To be sure, THX strongly recommends the use of dipolar surrounds for side wall placement. This new monopole spec coincides with the development of the next item –the ASA or Advanced Speaker Array.

4) The ASA (Advanced Speaker Array) is both acoustic and electronic as it requires a very specific loudspeaker configuration and complementary electronic processing.

From the speaker placement perspective, ASA makes use of the standard LCR "screen speaker" array as well as dipolar surrounds. In addition, ASA positions two additional monopole speakers side-by-side on the rear wall as follows:



On the processor end, ASA analyzes the incoming signal to determine the best way of presenting the material through the loudspeakers in the home theater room. Extensive DSP-produced cross-feeding of surround information emulates those so-called "spreader circuits" or "simulated stereo" effects we've all heard. The difference here is that far more sophisticated psychoacoustic modeling lies at the heart of this process.

The advantage of the full ASA package (speaker positioning and processing) lies in the fact that one loudspeaker array serves for both movies and music. This is no small accomplishment. Although ASA does not equal dedicated systems specifically configured for either movies or music, it does solve a major problem for many potential customers. If "automatic enjoyment" is high on your priority list, you might well audition a THX Ultra2 system.

ASA isn't mandatory, however. If you wish, you can forgo ASA in favor of a system with dipoles on the side and two monopole speakers in back. If you're watching non-EX movies, turn the back speakers off. When listening to multi-channel music, turn the side dipoles off. When watching an EX film, use all four surround speakers. Here, the back monopoles do a credible if not perfect job delivering EX. For non-critical listening, use all four surround speakers just for the thrill!

THE BOTTOM LINE

With this information, you're now in a much better position to make a judgment as to what THX offers.

For some people, the THX approach definitely results in increased enjoyment. Others see it as an interesting collection of different technologies that makes surprisingly little difference in their involvement with their favorite sources.

THX, particularly in its current "state of the art" Ultra2 iteration, will definitely cost more than a competently engineered non-THX system. Is it worth the money? That's a relative judgment we can't make for you. The best answer lies in what your ears tell you.

But remember that in the final analysis, room acoustics and proper set-up and calibration are far, *far* more audible than the gear assuming, of course, that you've chosen any of the better quality products in the market.

PREVIEW

DVD (correctly called Digital Versatile Disc, by the way) is the single hottest product category in consumer electronics today. In fact, recent surveys show that one in three U.S. households now have a DVD player as part of their home entertainment system. That's pretty impressive for a format that really hasn't been around all that long.

The main reason for this popularity is that DVD truly is versatile. DVDs can hold full-length movies, high definition audio, computer programs and data files, and much more. In addition, we'll soon see a plethora of (incompatible, of course!) formats for recordable DVD.

This section of the Rotel Home Theater and Hi Fi Encyclopedia will focus on the technology behind DVD. We'll take a look at how the format came to be and what all the buzz is really about.

Enjoy.

A BIT OF HISTORY

In The Lab

DVD development began just after the CD was commercially introduced in the mid-1980s. But It wasn't just one product development project hidden away in one company's labs. Almost all the major companies with any real knowledge of optical disc data storage were working on *something*.

The chief goal of these research programs was greater data storage. That's because video signals require far more storage capacity than audio signals. One avenue of inquiry involved the so-called "blue laser" that generated a short-wavelength beam of coherent light compared to the conventional "red laser" used in CD players. The blue laser was attractive because it could focus on far smaller areas and that meant greater data storage potential, a real necessity if the next generation of optical discs would be video-capable. However, the blue lasers of the day proved impractical for many reasons and research efforts dwindled.

Enter Hollywood

Other laboratory efforts were only marginally successful and no commercially feasible proposal came out of the labs. However, that didn't stop the film industry from putting together a list of its own requirements for the next generation video disc. Among other items, the film community asked for ("demanded" is a somewhat more accurate description) the following through an organization called the Hollywood Digital Video Disc Advisory Group:

1) Playing time of at least 135 minutes per side (the nominal length of a feature film)
2) Copy protection to prevent piracy
3) Better picture quality than laserdisc

4) Surround sound capability
5) Multiple soundtracks in different languages
6) Multiple aspect ratio support
7) A multiple version capacity with optional "parental lockout" to protect the eyes and minds of the innocent from seeing or hearing things perhaps too raw for their delicate natures.

In response to these "requests," the hardware community responded with two technical proposals – two formats that were different and (of course) incompatible.

The War Heats Up

Sony and Philips, anxious to preserve the status and cash flow resulting from their co-development of the CD, answered with the Multi-Media Compact Disc (MMCD) in December, 1994. A month later, Toshiba and its allies announced the SD-DVD (Super Density Digital Video Disc).

According to its proponents, the MMCD would be a single-sided disc with 3.7 gigabytes of total storage capacity. The SD alliance proposed a two-sided disc, each side with a 5 gigabyte capacity. Shortly after, Sony/Philips upped the ante with a dual layer (but still single sided) proposal totaling 7.4 gigabytes.

Computer Industry Concerns

As you might expect given the companies involved, neither proposal thrilled the computer industry. In retaliation for this slight, five important computer-related companies (Apple, Microsoft, IBM, Compaq, and Hewlett Packard) threatened a boycott. Their concerns included:

1) Absolute insistence on a single proposal to satisfy both home entertainment and computer applications.
2) A common disc format *and* a common file format for both types of use.
3) The new format must be backwards compatible with CDs and CD-ROMs.
4) The new format must provide for recordable discs.
5) Drives and discs should be cost-competitive with the CD-ROM format
6) There should be no cartridge to hold the disc itself.
7) The new format should provide at least the same level of data integrity as did CD-ROM.
8) High data capacity to protect against obsolescence.

At Last, Standards!

Although neither the MMCD nor the SD proponents wanted to budge, pressures from the computer industry and other special interest groups prevailed. By December, 1995, the broad brushstrokes for a single proposal were in place.

Eventually, specific standards emerged and were promulgated in a series of "Books."

Book A:	DVD-ROM
Book B:	DVD-Video
Book C:	DVD-Audio
Book D:	DVD-R
Book E:	DVD-RAM

Other "unbooked" formats (DVD-RW and DVD+RW, to name the two most prominent) are mostly variants on DVD-R. Although they will be important as we move past CD-R and CD-RW into the world of DVD's high-density data storage capacity, they probably won't affect the consumer electronics industry – except for the confusion their presence causes – for a few years yet.

Publication of the standards, though, was only a small step towards DVD's appearance as a commercial reality.

One glitch was that early DVD player prototypes couldn't read CD-R discs because the dyes used in a CD-R wouldn't reflect the shorter DVD laser wavelength. Sony and Philips, again trying to safeguard their CD royalty steam, proposed a new audio-for-DVD proposal called DSD or Direct Stream Digital. (This eventually came to market as SACD or Super Audio Compact Disc.)

Three Stumbling Blocks

Although some thought that 1996 would see the first DVD-V players, several unsettled technical and legal issues prevented this. By June, the commercial introduction of DVD was seriously behind schedule.

Copyright Issues

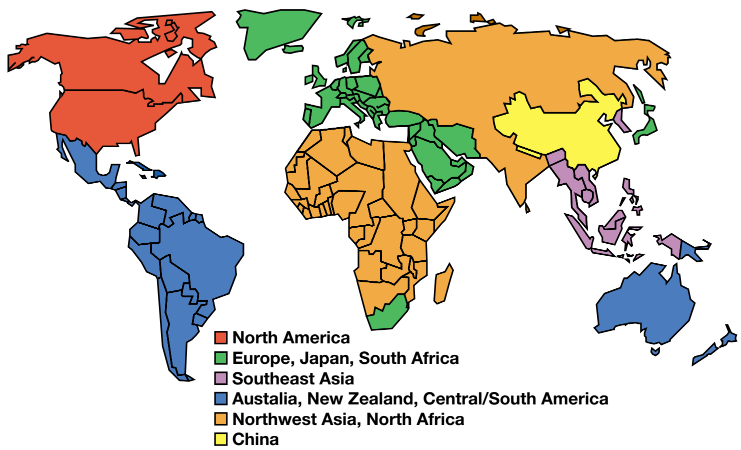
The biggest problem was the copyright question. The movie industry was convinced that a digital disc was simply an invitation to widespread piracy and lobbied for content protection. Two other issues, regional coding and software availability, were also unresolved.

CEMA (the Consumer Electronics Manufacturers' Association – now simply called the Consumer Electronics Association or CEA) and the MPAA (Motion Picture Association of America) proposed copyright legislation to which the computer industry strongly objected. But it wasn't until that October that the first real technical answer, jointly developed by Matsushita and Toshiba, was announced by the Copy Protection Technical Working Group.

Regional Coding

The regional coding issue was another challenge. Regional coding, by the way, simply meant that players sold in Europe, for example, wouldn't play discs made for distribution in the USA. Hollywood saw this as the only way to maintain the staggered distribution schedule for major films it had used for decades. Occasionally, release dates were so spread out that a movie could appear here on DVD before it even hit theaters overseas. To avoid problems, the DVD Forum split the world into six zones. Players sold in each zone would be set to play discs intended for that zone only. (Coding was optional for the discs, though. In other words, a studio could opt *not* to include a restrictive regional code on the disc itself.)

Here are the different regional coding zones:



All DVD players – and some discs – are coded so that only discs carrying the same regional code as the player would produce picture and sound.

Region	Area Covered
1	U.S. (including U.S. Territories), Canada
2	Europe, Japan, South Africa, Middle East (including Egypt)
3	Southeast Asia including Hong Kong
4	Mexico, Central America, Caribbean, South America, Australia, New Zealand, Pacific Islands,
5	Former Soviet Union, Indian subcontinent, Africa, North Korea, Mongolia
6	China
7	Reserved
8	Special international venues (airplanes, cruise ships, etc.)

For all the hoopla, regional coding has proven to be far less than the sum of its parts.

First, information on how to defeat regional code restrictions in the players themselves began to appear on the Internet almost as soon as the players were offered for sale. Some "fixes" involved nothing more than moving certain jumpers on a circuit board or, even easier, pressing certain key combinations on the remote. In addition, some manufacturers simply set their players for "Region 0" which, in effect, meant that that player would play a disc regardless of the disc's regional code.

Software Availability

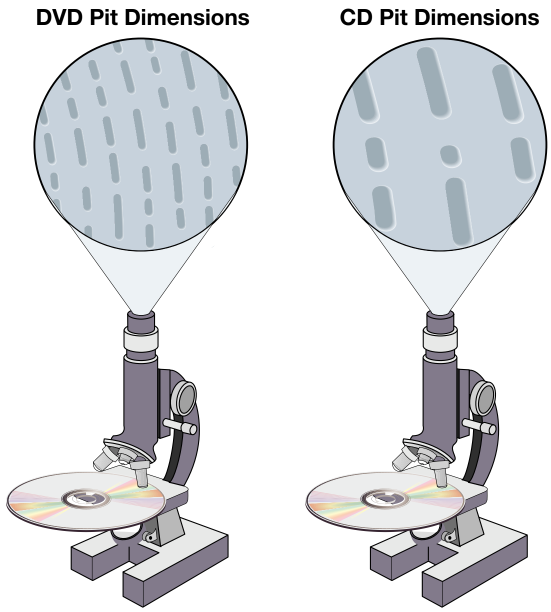
The CES (Consumer Electronic Show) of January, 1997 marked the end of the studio holdout. Six major studios (New Line Home Video, Polygram, Warner, and Columbia/TriStar among them) announced 60+ titles for March availability. On the hardware side, more than 16 consumer electronics hardware manufacturers announced specific products and 40 computer companies announced that DVD-ROM drive would be available momentarily.

February saw the first players. Discs were much scarcer. Warner announced a 7-city test market release with a national rollout to follow "soon thereafter." .

Despite the delays and confusion, consumers stepped up to the plate in a big way. More than 30,000 players and 50,000 discs were sold in just two months!

DVD DISCS

A DVD disc has, as we've already mentioned, phenomenal storage capacity. And that capacity is sorely needed as it takes about 2 gigabytes of data for an hour of full-motion video. As a point of reference, that's more than three times the data capacity of a single CD. And this doesn't include audio data or information needed for error correction, menus, etc.



Smaller pits, closer spacing, and a tighter pit "spiral" let the DVD carry much more information than a CD.

How did the DVD pack so much data on a disc? For one thing, DVDs store information more efficiently. A DVD laser generates a shorter wavelength beam than does its CD cousin. As a result, the DVD player can focus on a much smaller area. This means that the pits carrying the information can be smaller and that the pit spiral can be tighter. The dimensions here are truly miniscule. In fact, if we straightened out the pit spiral on one DVD layer, we'd have a line almost 7½ miles long!

And let's not forget the DVD specifications allow both double-sided and dual-layer discs, something simply impossible under the CD standard. In fact, if a content provider wanted to, they could produce a disc with two layers on each side. Or a disc with two layers on one side and only one layer on the other side! We'll provide more details about possible disc configurations shortly.

> ***Note: All the data figures we've mentioned so far pertain to 12 cm (4.7") discs. You may not know that the DVD specification also allows 8 cm (3") discs. These are the same size as those poorly received "CD singles" of a few years ago.***

Comparing CDs and DVDs

Here's a quick comparison of the similarities and differences between CDs and DVDs.

Specification	CD	DVD	Comments
Physical Dimensions			
Diameter	12 cm (4.7") or 8 cm (3")	12 cm (4.7") or 8 cm (3")	The same
Thickness	12 mm	12 mm (2 x 0.6)	All DVDs are made of two thin discs glued together.
Mass (12 cm disc)	14 to 33 grams	13 to 20 grams	CDs are heavier.
Data Storage			
Data area diameter	50 to 116 mm	48 to 116 mm	DVDs have more data area
Pit length	0.83 – 3.1 µm	0.4 – 1.9 µm (SL) 0.44 – 2.05 µm (DL)	DVDs' significantly smaller pit dimensions mean higher data density
Spiral track pitch	1.6 µm	0.74 µm	DVDs' pit spiral is tighter (2x that of a CD) for increased storage capability
Total capacity	Approx. 0.650 G bytes (650 megabytes)	1.4 – 8.0 G bytes/ side (2 sides possible)	DVDs have far more storage capacity
Miscellaneous			
Laser wavelength	780 nm	635 or 650 nm	DVD laser has shorter wavelength
Rotation speed	500 (inner) to 200 (outer) rpm	1600 (inner) to 570 (outer) rpm	DVDs spin 300% faster than CDs
Error Correction Overhead	23 – 34%	13%	DVD error correction much more efficient - needs fewer "redundancy bits"
Correctable error	2.5 mm	At least 6 mm	DVD error correction is far more effective

Disc Types

As we mentioned earlier, the DVD specs allow quite a number of disc types. Here is a quick chart to help you sort things out.

Official Designation	**Diameter**	**Sides/Layers**	**Data Capacity**	**Play Time**
DVD-5	12 cm (4.7")	One/One (SS/SL)	4.38 gigabytes (4.7 G)	2 hours +
DVD-9	"	One/Two (SS/DL)	7.95 gigabytes (8.5 G)	Approx. 4 hours
DVD-10	"	Two/One (DS/SL)	8.75 gigabytes (9.4 G)	Approx. 4.5 hours
DVD-14	"	Two/Mixed DS/ML (Two layers on one side, one on the other)	12.33 gigabytes (13.24 G)	Approx. 6.5 hours
DVD-18	"	Two/Two DS/DL	14.8 gigabytes (17 G)	8+ hours
DVD-1	8 cm (3")	One/One	1.33 gigabytes (1.4 G)	½ hour +
DVD-2	"	One/Two	2.48 gigabytes (2.7 G)	Approx. 1.3 hours
DVD-3	"	Two/One	2.72 gigabytes (2.9 G)	Approx. 1.4 hours
DVD-4	"	Two/Two	4.95 gigabytes (5.3 G)	Approx. 2.5 hours

"SS/SL" simply means single side/single layer. SS/DL, then, means "single side/double layer" and so on.

Keep the following in mind as you review this information.

First, we haven't included any data on the four competing recordable DVD formats (DVD-R, DVD-RAM, DVD-RW, DVD+RW). Specifications, particularly those relating to data storage capacity and compatibility, are somewhat in flux although most are now standardizing on 4.7 G capacity. We expect this situation to straighten itself out – and some of the competing formats to conveniently disappear – in the next year or so.

Next, we won't spend much time on the 3" discs as they are in very limited use at present. This, however, may change, particularly if software manufacturers choose this less expensive format for wide-scale program distribution. However, we don't expect to see the 3" discs become any more of a factor for consumer electronics products than the (in)famous "CD single" did several years ago.

Last, and of somewhat greater complexity, is the fact that the chart has two data capacity measurements per disc, one in gigabytes, the other identified with a "G." This is because computer people and other scientists use a different notation for expressing data capacity.

TWO WAYS OF COUNTING

The computer folks use "megabyte" as a basic unit of measurement: One megabyte = 1,024 bytes (or 210). However, the metric system, if strictly interpreted, would define a "megabyte" as "1,000 bytes." The initial discrepancy, only 24 bytes when we begin, quickly grows as the amount of data increases.

If we take a DVD-5 disc as our example, we can illustrate this difference easily. The single layer holds 4.7 billion bytes (or 4.7 G bytes) of information according to metric convention. However, if we expressed the same capacity in the language best understood by computer people, we'd end up with 4.38 megabytes because of the "1 megabyte = 1,024 bytes" equation. Thus, a DVD-18 (ds/dl) disc holds either 17 billion bytes (17 G bytes) or 15.9 gigabytes depending on which contingent of scientists you were talking to.

Here are some facts to remember:

1) An ss/sl (DVD-5) disc holds almost **7 times** the data of a conventional CD.

   A CD's capacity is approximately 650 megabytes (or 0.65 gigabytes). The ss/sl disc holds 4.38 gigabytes. Simple division produces 6.74. OK, it isn't 7. But it's close.

2) A ds/dl (DVD-18) disc holds more than **24 times** the data of a conventional CD.

   Disc types (DVD-5, DVD-10, etc.) describe only physical structure and data capacity. These designations have nothing to do with what type of data we put on the disc. You might find movies on DVD-5 or DVD-10 discs or computer data on a DVD-18 disc.

   From a manufacturing point of view, the easiest – and the cheapest – disc to make is the DVD-5 (or SS/SL). With playing time just over 2 hours, it's more than enough for most feature-length movies. Many movies also come on a

DVD-10 (DS/SL) disc that can hold a "pan and scan" version on one side and a widescreen version on the other. DVD-18 (DS/DL) discs will probably be the last to arrive in quantity since they are far more complex and costly to manufacturer.

A Quick Cost Comparison

One of the reasons DVDs cost more than CDs are the so-called "authoring" costs associated with making a DVD

These include compressing the video and audio portions of the program, creating and integrating the menus and other control information, and, finally, "mixing" everything into a single datastream prior to final encoding.

Typical compression charges approach $100/minute for video and $20+/minute for audio. Additional charges – subtitling, final formatting, testing, etc. – add to the tab. So how much will it cost to prepare a two-hour movie for DVD? Probably not less than $20,000! Mastering a CD will cost about $1,000. Duplicating DVDs runs about $1.60 per disc compared to the $0.50 charge per CD.

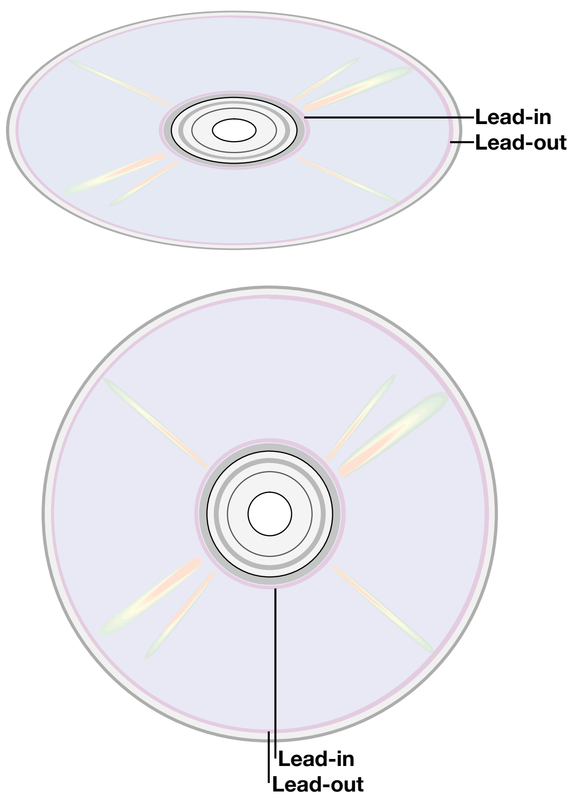
Some Scanning Mechanics

As do CD players, DVD players *usually* play discs from the inside out. (Notice the *usually*.) DVD players, though, spin DVD discs about three times faster than a CD player turns a CD. (DVD players spin CDs at CD-standard speeds in case you were wondering.) As with CDs, a DVD always turns counterclockwise when viewed from the laser's perspective.

The rotational speed (and this is true for both CDs and DVDs) decreases as players scan further and further from the center of the disc to the outer diameter. This keeps the data flowing at a constant rate. Technically, this variable rotation speed technique is called *constant linear velocity*. If the speed of rotation remained the same regardless of where on the disc surface the player was scanning, we'd call it *constant angular velocity*.

In most cases, the player first scans the "lead-in" area located at the center of the disc. This "lead in" area contains, among other things, information about the physical disc type, data formats, etc. When the player registers this information, it can then read the rest of the data until it encounters the "lead out" area at the disc's outer edge. In most cases, the player just switches the laser off and returns the assembly to the starting point, and goes into "standby" mode.

Dual layer discs often present an exception to this general scanning strategy. But not always. That's because there are two types of dual-layer discs – OTP (opposite track path) and PTP (parallel track path).



The player scans a single-layer side from the "lead in" area at the disc's center to the "lead out" area at the disc's outer diameter.

OPT discs comprise the vast majority of dual-layer discs. Here, the laser scans the first layer (or "Layer 0" in tech-speak) from the inside out and then refocuses on the other layer (Layer 1) to scan it from the *outside in*. This minimizes any gaps in playback and is the preferred method for longer movies released on dual-layer discs.

The far less common PTP disc is used for special applications when the player needs to switch quickly from layer to layer and back again.

A Note on DVD-ROM and Recordable DVDs

Although we see DVD primarily as an entertainment medium, the reality is somewhat different. In fact, industry sales figures show us that computer-specific DVD-ROM drives are outselling stand- alone consumer machines. As recordable DVD becomes more affordable and less confusing (i.e., competing formats disappear), this balance will only increase.

For that reason, you need to know a bit about those formats so we'll present some very brief comments here. Please understand that this scene is changing on an almost daily basis and so we caution you to check the web for the latest information before making any recommendations to your customers.

With that caution, here's some of what you need to know:

DVD-ROM (DVD-Read Only Memory)

This is the DVD version of CD-ROM and is intended for computer applications. DVD-ROM's major advantage is its storage density. For example, the most recent version of Microsoft's Encarta encyclopedia comes on 6 CD-ROMs – or 1 DVD-ROM!

DVD-ROM drives read CDs and CD-ROMs so there're no backwards-compatibility issues to be concerned about. One thing you need to remember, though, is that some first-generation DVD-ROM drives can not read CD-R (recordable) discs because the dyes used in these CD-Rs will not reflect the DVD-ROM drive's short-wavelength laser. They will also probably have trouble with CD-RW (CD-rewritable) discs because a CD-RW reflects less of a laser's light than other disc types.

DVD-R (DVD Recordable)

This is the first recordable DVD format and uses the same organic dye polymer chemistry used by CD-Rs. Like a CD-R, a DVD-R is a write-once disc. A mistake means you've got another polycarbonate coaster to put under whatever you're imbibing at the time.

Designed primarily for authoring, DVD-R discs are compatible with almost all DVD drives and players. DVD-Rs come in two versions: DVD-R(A) for professional use and

DVD-R(G) for consumer use. DVD-R(A) discs won't work in DVD-R(G) drives.

Although first-generation DVD-R discs have only 3.95 G bytes of storage capacity, more recent (and far more numerous now) second-generation discs and drives handle 4.7 G bytes.

DVD-RW (DVD-Rewritable)

Developed and championed primarily by Pioneer, DVD-RW uses phase-change erasable material similar to that used in the CD-RW and is compatible with most DVD drives and players. (Be a bit careful here – some drives will be confused by DVD-RW's lower reflectivity and think they're reading a dual layer disc. They aren't – DVD-RW discs are single layer only.)

DVD-RW discs will provide approximately 1,000 record/erase cycles although media improvements will probably increase this.

Data capacity is 4.7 G bytes.

Although DVD-RW drives and media were extraordinarily expensive when first introduced, prices have fallen to the point that this is a commercially viable format now.

DVD-RAM

Developed by Hitachi and backed primarily by Panasonic today, DVD-RAM uses phase-change technology but, like the MiniDisc, uses some MO (magneto-optical) features, too. One strong advantage is that a DVD-RAM disc may last through more than 100,000 record/erase cycles. DVD-RAM uses a caddy to hold discs when recording. Single-sided discs can be played back without the caddy, double-sided discs cannot.

There are both physical and data file format incompatibilities that make DVD-RAM incompatible with conventional DVD players. So you can't record from your camcorder, for example, onto a DVD-RAM disc and then play that disc in a conventional DVD-V player.

Some recent DVD-RAM drives also produce DVD-Rs but there are still compatibility issues.

DVD+RW

Backed by Philips, Sony, Hewlett-Packard, Dell Computer, and others, DVD+RW combines DVD and CD-RW technology. Compatibility is a big plus as even first-generation DVD+RW drives read CDs, CD-ROMs, and DVD-ROMs and current drives also read DVD-R and DVD-RW discs. A DVD-RW drive will not read a DVD-RAM disc.

DVD+R is simply a "record once" version of DVD-RW.

Although DVD+RW's data capacity was first pegged at 3 G bytes, improvements have increased that to 4.7 G bytes.

DVD-VIDEO

Before we get into the technical matters, let's begin by reminding ourselves *why* DVD-V (the accepted abbreviation for DVD-Video) has become as popular as it has. The following list contains some of these reasons:

1) A DVD-Video disc Is compact, easy to handle, store, and transport; replication is comparatively inexpensive; players can be small.
2) The format is convenient: Near-instant access to tracks and chapters.
3) One disc can hold from 2 to 8+ hours of high-quality digital video.
4) The format allows multi-channel sound capability
5) Each disc can hold up to 8 separate tracks of digital audio (Dolby Digital, multiple languages, etc.)
6) DVD-V discs play widescreen movies on both standard (4:3 aspect ration) or widescreen (16:9 aspect ratio) TVs.
7) The discs have navigation "menus" so viewers can find different scenes, etc.
8) DVD-V discs are exceptionally durable and surprisingly heat resistant.
9) Discs may contain multi-lingual text keyed to the available soundtrack language choices provided. The text may include subtitles, lyrics, credits, cast information, crew identification, etc.
10) The format also allows automatic "seamless" video branching so one disc can show different story lines or, if parental controls are activated, skip potentially offensive scenes.
11) Each disc has the capacity to show up to 9 different camera angles for each scene (if they were filmed in the first place, of course) so the viewer can select the angle of greatest interest.
12) The disc's non-magnetic construction means that magnetic fields won't affect the disc's contents. This makes storage easier.

Note: Very few discs contain all of these features. There are no technical barriers to a "loaded" disc but there are severe financial ones – each feature must be specially "authored" into the master datastream. As we've already seen, authoring is expensive.

A Short Primer on Video Technology

Although we're going to slide through this area rather quickly (and advise you to do some research if you're at all interested), there are some facets of video you need to know in order to understand DVD more fully.

Those areas are:

1) Interlaced vs. progressive scanning
2) NTSC, PAL, and SECAM formats
3) Aspect ratios

Interlaced vs. Progressive Scanning

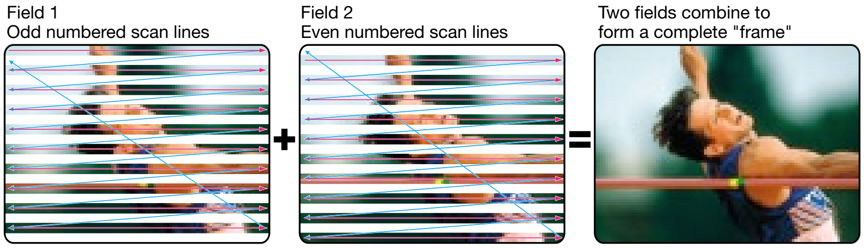
Engineers developing television faced a major hurdle. They had to deliver a flicker-free image because most people are very sensitive to any discontinuities as we view a "moving" image.

The movie industry had already dealt with this problem when it selected a "frame rate" of 24 frames per second (fps). This rate meant that a sequence of still images would appear to be a continuously moving one. This is a manifestation of something known among perceptual psychologists as the "flicker fusion frequency."

After deciding on a frame rate of 30, TV engineers ran into a hardware blockade. TV tubes (more properly known as CRTs or "cathode ray tubes") and cameras of the day simply couldn't reproduce 30 frames a second without a substantial decrease in resolution.

A TV picture, as you probably know, is composed of many thin lines formed by an electron beam that quickly scans the inside surface of the picture tube. The beam is generated by electron "guns" (or a single "gun") located at the rear of the picture tube. From the viewer's perspective, the beam starts its trace at the upper left hand corner of the tube, scans horizontally across to the right side, jumps back to the left, drops just a bit, and begins to scan again. You'll find more details about this in just a few paragraphs.

Engineers quickly evolved a solution by which they divided each frame into two fields, each one composed of alternate scanning lines. Thus, the first field would contain only the *odd* scan lines, while the next field would contain only the *even* scan lines. The engineers counted on our comparatively "long lasting" visual memory and the comparatively "slow" phosphors of the day to fuse those two fields into a single, higher definition frame. That's why we call a conventional TV picture "interlaced."



The number of scan lines shown is only a schematic representation. Actual screens have more than 20 times the scan lines shown here.

Conventional interlaced television traces half of an image on the screen and then traces the second half. Visual memory and slowly decaying phosphors fuse these into a single image.

The drawing above simply shows the concept of an interlaced picture. The actual scale is much different. The current United States TV standard (it's used elsewhere, too) is called NTSC or National Television Systems Committee). NTSC frames are composed of 525 "scan lines" (each equivalent to a *vertical* line of resolution). Each frame is actually made up of two different fields, each one with 212.5 lines. Given a frame rate of 30 fps, it doesn't take higher mathematics to figure out that the field rate is double that - or 60 fps.

The number of scan lines shown is only a schematic representation. Actual screens have more than 40 times the scan lines shown here.

Progressively scanned images result from a single electron beam scan.

The Progressive Edge?

Progressive scanning is quite different. Here, images have no individual fields – each frame is traced onto the display device's viewing surface with a single even-higher-speed pass of the electron beam. That's the major reason why progressive scanned images are better than interlaced images. They can convey shimmer-less, ripple-free pictures without the artifacts we usually see in interlaced images. Progressive scanning, though, carries a price. It requires twice the bandwidth of an interlace signal.

Sources

The reason the differences between interlaced and progressively scanned images are important is that DVD stores information in a form that is neither purely interlaced nor purely progressive. (Don't ask why unless you want to hear a long, *very* technical, perhaps even mind-numbing treatise on such obscure topics as "macroblock efficiencies.")

The reason for this is that DVD was conceived as a format suitable for both types of images. If you look at the most prevalent sources for material to be captured on DVD, you'll see why.

Until very recently, most video cameras, even professional ones, processed interlaced images. Those images – virtually every TV show, home videos, etc. – comprise an important source of material for DVD. Film, on the other hand, is an inherently progressive format. Although the frame rates are different (24 frames per second for film, 30 per second for video) and these differences pose other challenges ("3-2 pulldown" for example), DVD technology still allows retention of much of the detail inherent in an "originally progressive" film.

NTSC, PAL, and SECAM Formats

NTSC is *not* the only standard used today. PAL and SECAM standards (used mostly in Europe and Asia) also exist.

Format	Used in . . .	Lines of Vertical Resolution (Total/Visible)	Frame/Field Rate
NTSC	North America, Japan, etc.	525/480	30/60
PAL	Western Europe (except France), Hong Kong, mainland China, etc.	625/576	25/50
SECAM	France (primarily) and some European countries formerly in the "Eastern block"	625/576	25/50

Some notes:

NTSC is sometimes referred to as "never the same color." That's because the system sometimes produces, er, "eccentric" results. But it's the one we've got. At least until DTV gets here.

PAL (or Phase Alternate Line) is used mainly in Europe. Actually, PAL comes in several variants distinguished mainly by different color subcarrier frequencies. If you're from the U.S., PAL really means "picture always lousy." Europeans, of course, say PAL means "perfect at last."

SECAM (sequential couleur avec memorie – "sequential color with memory") is a PAL variant used primarily by the French and in some former Eastern Block countries.

As you see, these formats are all "interlaced" – they use two fields per frame to reconstruct a picture with reasonable resolution.

THE WAR OF THE DISCS: NTSC VS. PAL/SECAM

To add to the confusion, differences between NTSC and PAL formats extend to DVD discs also.

Although all DVDs contain MPEG video data stored in digital form, this data is formatted for either NTSC or PAL systems. Thus, there are two kinds of DVDs: NTSC and PAL. Some players only play NTSC discs, others play PAL and NTSC discs. (Both NTSC and PAL discs may carry Regional Codes also.)

Fortunately, at least for those who live in countries using the NTSC standard, this may never be a issue as more than 95% of all DVD titles are NTSC-encoded.

However, that means that 5% of all discs are PAL-encoded. If you encounter one, will you be able to enjoy it?

That depends on a number of factors: the disc itself, the player, and the display device on which you watch – or try to watch! – the image.

The following table will tell you what you need to know. (Unless otherwise noted, all comments about PAL discs, players, or display devices refer to PAL/SECAM-capable units.)

Disc Type	Player Type	Display Type	Comments
NTSC	NTSC-only	NTSC-only	Compatible.
"	"	PAL-only	No picture. PAL-standard display unable to use NTSC video data.
"	"	Multi-standard (NTSC or PAL)	Compatible as long as display's NTSC format chosen.
"	Multi-standard (NTSC/PAL)	NTSC-only	Video information viewable. Some players provide switch-selectable "psuedo-PAL" or true NTSC output. .
"	"	PAL-only	Video information viewable by choosing player's "Psuedo-PAL" output. Player may not decode MPEG audio
"	"	Multi-standard (NTSC or PAL)	Choose player's "Psuedo-PAL" output and PAL input on display or On players with full NTSC format conversion, use NTSC output from player and NTSC input on display. (Preferred.)
PAL	NTSC-only	NTSC-only	Player cannot read video information on disc. Player will not decode MPEG audio.
"	"	PAL-only	Same as above
"	Multi-standard (NTSC/PAL)	NTSC-only	On players with full NTSC format conversion, use NTSC output from player to display. Player's "pseudo-PAL" output useless as display does not accept input.
"	"	PAL-only	Compatible
	"	Multi-standard (NTSC or PAL)	Use PAL output on player to PAL input on display.

In summary, here are some points to remember:

- *Most NTSC players can't play PAL discs.*
- *A few (very few) NTSC players can convert PAL to NTSC. External converter boxes are also available.*
- *All DVD players sold in PAL countries play both kinds of discs.*
- *Not all NTSC players play MPEG audio tracks but most studios put Dolby Digital audio tracks on their PAL discs instead of MPEG audio tracks.*
- *All PAL DVD players play Dolby Digital audio tracks*
- *Multi-standard players partially convert an NTSC video disc's data to a "psuedo-PAL" output that combines 60-Hz fields and PAL's 4.43 MHz color subcarrier. This format outputs a video signal at the 525/60 NTSC scan rate. (Most modern PAL TVs accommodate this signal.)*
- *Some players have a switch to choose 60-Hz "psuedo-PAL" or true NTSC output when playing NTSC discs.*
- *Some standards-converting PAL players convert NTSC discs to standard (non-"psuedo") PAL output. This is expensive to implement properly and is rarely done. Instead, most of these players use "Psuedo-PAL" outputs instead. (Psuedo-PAL outputs usually produce better picture than improperly implemented standard PAL conversion.)*

A Question of Resolution

DVD brings us more detailed images than other consumer video formats available today. Here's a quick comparison:

DVD:	(Up to) 540	lines of horizontal resolution
Laser Disc:	425	" " " "
VHS:	240	" " " "

Notice that we've based this quick comparison on lines of horizontal resolution. This may be initially confusing as we've already stated that an NTSC TV is capable of displaying only 480 lines. If this is true, aren't those other 60 lines (the difference between 540 and 480) superfluous?

The answer is no. The reason is that there is a major difference between the 480 line specification and the 540 line specification.

The 480 figure is the total of *visible* scan lines you can see in an NTSC picture. It is a measurement of *vertical* (up and down) resolution. (Incidentally, we never see all 525 scan lines that make an NTSC picture. A large number of lines carry sync information and other data needed to keep the picture stable on the screen.)

The "up to 540 lines" DVD spec (and the 425 and 240 specs mentioned in the same comparison) measure *horizontal* (side-to-side or edge-to-edge) resolution. These numbers tell you how many *pixels* (picture elements) make up each scan line, not how many scan lines make up a full frame. The more horizontal pixels, the higher the resolution. It's that simple.

That's one reason why DVD produces a much better picture than VHS tape (with its 240 or so pixels per scan line) or even a good broadcast source.

What's Your Aspect Ratio?

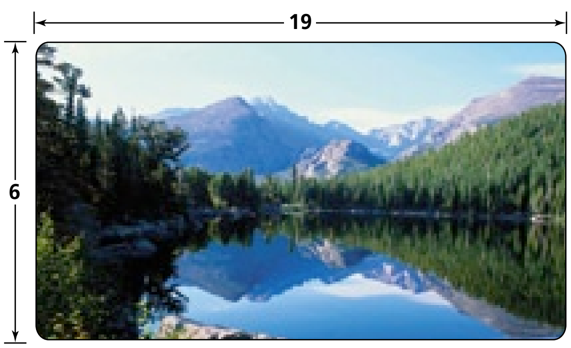
The final video element we'll deal with is something called aspect ratio.

Aspect ratio simply means the relationship between an image's height and width.

4:3 TV screen with aspect ratio clearly delineated

A conventional TV has what's called a 4:3 aspect ratio. This simply means that the image is always wider than it is high by a fixed ratio: 4 units (whatever length they may be) wide by 3 units high. As you see, every time you watch a conventional TV, pictures with this aspect ratio are almost square. NTSC, PAL, and SECAM TVs all use a 4:3 aspect ratio.

The standard aspect ration for HDTV (High Definition TV – the new all-digital standard) is much more like a movie screen – in other words, the image is much wider than it is high. Here, the aspect ratio is 16:9 or 16 units wide by 9 units high.



16:9 TV screenwith aspect ratio clearly delineated

Most DVDs carry 16:9 images that produce a "letterbox" effect when displayed on a conventional 4:3 TV.

A few (mostly older) movie discs may carry a 16:9 (widescreen) version on one side and a 4:3 version ("pan and scan") on the other.

Security Issues

Despite the technical *tour de force* DVD presented, the format wasn't welcomed by the Hollywood film community. Concerned that any form of digital replication would result in widespread piracy, studio executives and their lawyers pushed for assurances that discs would be "secure." .

16:9 image shown on a 4:3 screen with top and bottom black bards showing

As a result, today's DVDs *can* be protected by as many as *four* anti-copy technologies. Another technique called "watermarking" also enters the picture. These anti-copy provisions are not mandatory but you'd be hard pressed to find any unprotected discs today.

But who are the discs protected against? Although casual users (including most consumers) would find the safeguards daunting, there is frankly little assurance that technically savvy large-scale pirates will be dissuaded

In any case, the protective technologies available are:

1) **Macrovision:** This works in the analog domain to prevent dubbing a DVD onto videotape. Macrovision adds a "rapidly modulated colorburst signal" to the

composite and S-video outputs. Coupled with additional pulses in the vertical blanking signal, Macrovision upsets auto record settings and sync tracking in almost all VCRs.

Early DVD players with component video outputs may or may not have Macrovision protection. Recent ones, however, come equipped with a modified version that is compatible with component outputs.

2) **CGMS** (or Copy Generation Management System): This system works in both the analog and digital domains. Discs protected by CGMS contain "flags" in the digital datastream that tell a CGMS-equipped recording device if a copy (or a copy of a copy) is allowed.

3) **CSS (or Content Scrambling System): DVD** video files are coded so that they do not appear on the disc in a form that can be directly played back. Prior to playback, a DVD player must pass the data through a decryption circuit before the flowing circuitry can make any sense out of the data.

   As almost everyone knows, a decryption key (a short program called De-CSS) was published on the net in November, 1999. Although the authorities clamped down hard at the insistence of the film community, the fact that a Norwegian high school student was able to hack a tool that an entire industry took years to develop shows the inherent vulnerability of many digital encryption schemes.

4) **DCPS (or Digital Copy Protection System): This is the technology that was supposed to usher in the Age of Firewire. Although we are much closer today to implementing some form of DCPS, a lot of behind-the-scenes politics is clouding the technical issues.** DCPS addresses the problems inherent in providing direct component-to-component digital connections while still guarding against making theoretically perfect digital copies. A number of systems have been proposed.

   In general, DCPS is a "second level" encryption/decryption scheme that marks content with a "copy freely," a "copy once," or a "don't copy at all" flag. When successfully implemented, both the source component and the receiving component exchange encryption keys and authentication codes before establishing a "secure" transmission path.

   For example, if a DVD player encrypts audio and video information by using a special "do not copy at all" code that effectively restricts reception to only a television, it would prevent an "unauthorized" device (a digital VCR, for example) from hijacking the material.

5) **Watermarking**: While not, strictly speaking, a security technology in itself, watermarking is a technique that Hollywood hopes will help it track down the source of pirated material. Watermarking attempts to permanently embed digital noise in each digital video frame. Although proponents claim that watermarking is visually undetectable, the very robustness of the technique (it must be able to survive repeated digital-to-analog and analog-to-digital conversions as well as video processing) leaves some room for doubt as does the assurance that watermarked discs will be totally compatible with older players.

Audio-for-Video Formats on DVD

With some of the video basics covered, we now need to dive into audio on DVD. After all, Rotel is primarily an audio company!

As you might imagine, audio-on-DVD is a somewhat complicated topic. Before we get into the technical details, let's begin with some background.

Movie Soundtrack Audio

DVD-Video discs can carry three different audio formats: Linear PCM (1 to 8 channels), Dolby Digital (1 to 6 – read that 5.1 – channels), or 1 to 7.1 channels of compressed MPEG audio. (You won't find MPEG audio on NTSC-spec video discs. You may on PAL or SECAM discs.)

Content providers may choose an option (either DTS encoding from Digital Theater Systems or SDDS, Sony's Dynamic Digital Sound). Either may require an external decoder because DVD specs don't mandate either. Today, many DVD players include DTS decoders and many more provide "DTS pass-through" capability via a digital audio output to an external processor.

Here are some details on the mandatory formats:

Linear PCM (or LPCM)

Similar to the data structure on a conventional CD, linear PCM (short for Pulse Code Modulation) takes advantage of DVD's increased storage capacity by using higher data rates than does the CD. For example, rather than be restricted to the CD standard of 44,100 16-bit words per second, the audio data on a DVD may vary from 16- to 24-bits at sampling rates of either 48 or 96 kHz (48,000 or 96,000 words per second). In addition to straight two-channel information, these PCM tracks can also carry an LtRt Dolby Surround mix to an external decoder.

PCM data is uncompressed. Although theoretically capable of carrying up to eight channels of information, the maximum bit rate (6.144 Mbps) restricts PCM's multi-channel use to "less dense" data structures. In other words, you won't see 5.1 channels of 24-bit words at a 96 kHz sampling rate.

DVD players must support all the PCM formats. However, there is no rule telling manufacturers how. As a result, some DVD players "subsample" 96 kHz data down to 48 kHz. Even more simply cut up to 4 LSBs (Least Significant Bits) from a 24-bit datastream. In other words, they simply don't output all the information encoded onto a disc. (This is true mostly for earlier DVD players. More recent models have D/A converters fully capable of full "24/96" processing.)

Dolby Digital

Dolby Digital is the de facto standard for all DVDs, whether NTSC or PAL/SECAM. It allows up to 6 separate channels of audio information to be encoded on the disc. The DVD spec allows two different bit rates depending on what kind of sound quality is desired and how much room is left on the disc after the video is encoded. The usual practice is a data rate of 364 kbps (kilobits per second) although a 448 kpbs rate is possible. Many Dolby Digital decoders support bit rates up to 640 kpbs. However, there are few discs so encoded.

Dolby Digital is a lossy compression technique. That means some of the original data is permanently lost when converted to a Dolby Digital datastream. Although some observers have taken this literally, jumping to the conclusion that Dolby Digital is an inferior coding system, our experience is that this is definitely not the case.

Not all Dolby Digital datastreams carry 6 channels of information. The term Dolby Digital refers only to the way information is coded. A Dolby Digital soundtrack may be mono, stereo, surround-encoded, or full 5.1 or 6.1 discrete. Check the disc to make sure how many channels are encoded.

MPEG Audio

This format is similar to Dolby Digital in many ways. It uses lossy compression. It can support up to 7.1 channels, adding a Left-Center and Right-Center to the standard mix. MPEG bit rates vary from 32 to 912 kpbs, but the DVDs usually carry data at a conservative 384 kpbs rate. DVD supports both MPEG-1 (with a top data rate of 384 kpbs) and the potentially superior MPEG-2. DVD does not support MP3 or MPEG-2 AAC.

MPEG Audio was originally proposed as the primary audio standard for PAL/SECAM discs. However, Dolby Digital has largely supplanted it in common usage.

Here are the optional formats:

DTS

DTS is another lossy compression strategy. However, the data rate can be much higher (up to 1,536 kpbs) than most other lossy compression standards now in use. DTS advocates claim that the higher data rate is directly responsible for better sound. Although the argument is logical, it is not necessarily correct as the way a system compresses data is at least as important as the final bit rate of the data it creates.

When you realize that the data rate for DTS-encoded DVDs is half that system's initial rate (or 768 kpbs), the superiority arguments are not as strong as they might first appear.

Many early DVD players do not recognize a DTS datastream. Even today, you should look for at least "DTS pass-through" capability if not internal DTS decoding.

SDDS

Sony Dynamic Digital Sound is another optional multi-channel digital format. It uses the same lossy ATRAC compression technique used by Sony's MiniDisc. It supports both 5.1 and 7.1 channel output with data rates up to 1280 kbps. In the mind of most consumers, SDDS is a non-entity.

CAUTION – GET THE SOUND YOU REALLY WANT!

One of the prices we pay for DVD's extraordinary versatility is the confusion that often accompanies the large number of choices the format gives us.

Nothing illustrates this problem more than getting the right movie soundtrack off the disc and through our speakers.

Suppose you want to hear the Dolby Digital 5.1 soundtrack. After all, it's "the standard" so all you need to do is plop the disc in the player, press "Play" and enjoy the result, right?

Hah! We wish it were that easy.

The reality is that you might be listening to a two-channel Dolby Surround-encoded soundtrack decoded by your processor's Pro Logic circuitry – if you're lucky! If you're not, you might spend many frustrating minutes checking speaker connections and wondering where the center and surround channels went.

The answer to this dilemma is all in the menus – and you'll have to deal with three of them to make sure you're hearing what you want to hear.

The first is the disc's "audio" menu. Every disc is a bit different but the point of searching out this menu is to make sure that the format you want (Dolby Digital, DTS, or two-channel PCM, for example) is there on the disc in the first place. Once you've gotten to these choices, pick the one you want.

The second menu is the player setup menu. Getting to this one may be a bit arcane so read your owner's manual carefully. (The good news is that you'll probably only have to do this once.) After getting to the audio portion of the menu, you can usually select a default choice that the player will automatically revert to. Dolby

Digital is the safest as there MUST be a Dolby Digital soundtrack on all NTSC-compatible discs. (And most producers now include a Dolby Digital track on PAL discs, even though it's optional.)

Make sure you've also chosen the output you want your player to direct that soundtrack to. Some give you the option of 5.1 analog outputs or an optical or coaxial digital output.

Once you've done this, move on to your processor and make sure you've set it to receive the audio signal from the previously selected output on the DVD player. A growing number of processors now have "auto detect" circuits that monitor the incoming signal and automatically decode it properly, but you must make sure you've selected the proper soundtrack from the disc menu and have instructed the player to export it to the proper output in order to enjoy the marvels of surround sound.

Audio-Only Formats for DVD

The history of audio-for-audio's-sake on DVD is even more fractious. We'll briefly cover one format that's little more than a small footnote now. Then, we'll give a quick overlook to "set the stage" for in-depth discussions of DVD-A and SACD.

AAD (Advanced Audio Disc)

AADs are (or, more properly today, *were*) simply DVD-V discs without images. These discs usually had two channels of high-resolution "24t/96" PCM audio.

The AAD had no real future even when it was introduced as it merely paved the way for DVD-A. AADs have largely disappeared today although they are playable on any DVD-V machine (with the caveats mentioned in the last paragraph of the Linear PCM section above).

DVD-A and SACD: An Overview

The real reason these competing approaches to audio-on-DVD exist is not due to any immediately perceptible aural superiority of one over the other. It does, however, have its roots in corporate pride and finances.

Sony and Philips, recognizing that their patents – and their revenue stream – on the CD were running out, advanced a new single-bit approach called DSD (Direct Stream Digital) as the proposed standard for DVD audio. The DVD Forum, through its WG4 (Working Group 4, a technical committee charged with formalizing DVD's audio specifications) wasn't buying. The group's members, spearheaded by Sony and Philips' arch-rivals Toshiba and Matsushita, asked themselves why they should back a proposal that would benefit their competitors more than themselves. And they came to the obvious answer. They voted to use an extended menu of PCM-based bit structures and sampling rates they felt would advance their interests as well as be attractive to the music industry. In order to capitalize on the growing popularity of DVD in general, they called their proposal DVD-A (short for DVD-Audio).

Once the WG4 repudiated the Sony/Philips proposal, these companies decided to go ahead on their own and developed the SACD (or Super Audio Compact Disc).

The result is two different formats that use DVD's greatly increased data storage capability to carry high-resolution digital audio signals for playback in the home.

DVD-A is, as you'll soon see, a somewhat complicated mélange of different data structure choices. It has fairly extensive video capabilities, too. But it carries the sometimes frustrating overhead of a menu system that almost always mandates the use of a television to display the menu needed to play a disc.

SACD, on the other hand, traces its lineage more to the simplicity that helped make CD as successful as it is. SACD's video capabilities are somewhat anemic compared to a DVD-A but it doesn't require extensive menu navigation to get to the music either. However, it does not have the support of many of the biggest record companies either so its future is somewhat cloudy.

When first introduced, SACD was strictly a stereo format. Now, however, surround-capable players and surround-encoded discs are becoming more plentiful.

As of this writing (late summer, 2001), it is difficult to predict a winner in this "mini format war." Indeed, the term "format war" is itself misleading as both DVD-A discs and SACD discs will fit into the same transport! And, given the number of companies with ICs capable of decoding both DVD-A and SACD, true "omni-players" are restricted only by the politics of technology licensing. (That was definitely *not* the case when VHS and Beta butted heads together almost two decades ago – as anyone who tried to stuff a VHS tape into a Beta player will tell you!)

PREVIEW

If you read the last portion of Section 7 (DVD-A and SACD: An Overview), you already understand some of the politics of DVD-A and SACD. In this Section, we'll leave politics aside to look at the hard technicalities of the DVD-A format.

At the end of Section 9: SACD you'll find a handy comparison chart that will give you a clear picture of the difference between these important new digital audio formats.

WHAT IS DVD-A?

No question that DVD-A (short, of course, for DVD-Audio) is a hot topic today. Riding on the coattails of the super-successful DVD-V format, DVD-A is in the running to become the "next CD." Of course, SACD (Super Audio Compact Disc) is in there pitching too. Both DVD-A and SACD promise much better sound than CD but that isn't too surprising when you remember that basic CD technology is now over 20 years old!

However, DVD-A approaches the upscale sound market with a distinctly different perspective than does SACD. For one thing, DVD-A, despite the "audio" in its name, offers far more video capability. For another, it uses a very different type of digital data structure than SACD's single-bit DSD approach. And it's multi-channel capable right at the outset, a very different starting point than SACD's "stereo only" first-generation hardware and software.

Despite this, there's some dissention in the industry about what DVD-A *really* is. Depending on who you ask, DVD-A is either the most exciting digital audio format in years or a somewhat bloated grab-your-bucks scheme that's primarily designed to sell you yet another version of the Beatles' *White Album* and the machine you'll need to play it.

Indeed, there may be some truth in both perspectives. DVD-A is new. It is a technical *tour-de-force*. It does promise significantly better sound than CDs can deliver . . . at least in theory. (But, as you'll see, real world limitations make some of these theoretical advances a bit questionable.)

DVD-A also offers multi-channel audio capability from the outset. An advantage? To some, yes. But that depends on your preferences, the intentions of the music's composer, and the skill of the engineers who record and process the music in the first place. But it doesn't offer the same level of advances that instantly distinguished the CD from the LP in the early '80s either.

In short, there's a lot of good – and some strangeness – inherent in the format. This section will step you through the basics and share some experiences and opinions so you'll be able to weigh claims and counterclaims more objectively.

SHOULD I CARE?

Yes, you should. At least if you're interested in high fidelity, that is.

After all, DVD-A promises a substantial improvement in sound quality when compared to CD. Indeed, a spec-for-spec comparison can easily lead you to believe that you'll really hear an immediate and substantial difference.

One reason for DVD-A's "popularity" with the music industry is that it (the music industry) would like to get in one last "improvement" in our hands before we all move into the Age Of The Download when music will be distributed electronically and individual collections of discs or tapes will effectively cease to exist. (At least this is the dream.)

Remember that "legacy" music (albums or tracks that have already been recorded) is the single most profitable form of music a company can sell – and sell again. There's absolutely no need to go back into the studio to re-record it. (Of course, the master tapes have to be reformatted but that's a comparatively minor expense.)

BACKGROUND

As we've already mentioned, DVD-A grew from the work done originally to provide high-quality sound for DVD-V and takes advantage of DVD's substantial storage capability (up to 14.8 gigabytes on a double-sided, dual layer disc) to pack in a lot more data. By way of comparison, a CD holds "only" 650 megabytes. In theory, that extra data can mean better sound quality or longer playing times – or both.

WHAT MAKES A DVD-A DISC A DVD-A DISC?

In addition to purely physical differences (see the table entitled "Comparing CDs and DVDs" early in Section 9), there are other significant points of departure from CDs.

First, DVD-A discs can contain multi-channel music. This is a major difference from CD's "two channel only" limitation as multi-channel capability gives artists and producers a new tool to use. And engineers a whole new world to understand. Although we suspect that there will be some sonic strangeness at first, we're sure that this multi-channel capability will eventually provide a lot of enjoyment.

Physically, there's room for 8 separate PCM tracks – a fully discrete 5.1 surround *and* a 2 channel stereo version of the same music. In addition, a late addendum to the DVD-A spec now allows a content provider to include an *optional* Dolby Digital version of the same music.

THE COMPATIBILITY QUESTION: DVD-A DISCS IN DVD-V PLAYERS

Why two surround versions of the same music on one disc? That's the answer supplied by the DVD Forum's WG-4 when it was forcibly reminded that under the original proposal, millions of people who bought DVD-V machines wouldn't be able to play DVD-A discs at all!

Remember that a DVD-A disc is not a DVD-V disc. That's because the file formats and the disc's directory structure are just different enough to blind the microprocessors in a DVD-V machine to the very presence of DVD-A tracks. That's because DVD-A files are listed only in an "AUDIO_TS" directory that DVD-V players can't even find.

Also, remember that DVD-V machines weren't designed to handle 88.2, 176, or 192 kHz sampled data even though they will play a 44.1 kHz CD or other PCM files at 48 or 96 kHz sampling rates. MLP-encoded data in a DVD-V machine? No way at all.

There are two answers to this compatibility dilemma.

First, disc producers can (repeat can – but are not required to) make a DVD-A disc playable on a DVD-V machine by including a separate Dolby Digital version of the same material in the "DVD-Video" zone on the disc. Although some might complain that this wouldn't be the high-density PCM datastream that lies at the heart of DVD-A's claims to sonic superiority, it's a lot better than not hearing anything at all!

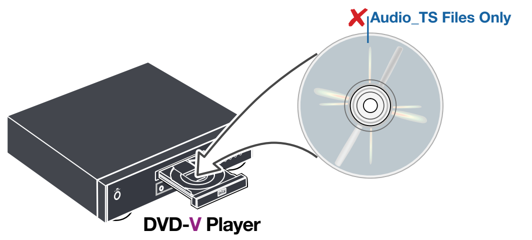
The second answer to this dilemma is the "omni-disc" playback machine we mentioned earlier. But that, at least as of this writing, is still a bit in the future.

From a technical perspective, there are four aspects of DVD-A discs we need to discuss. They are:

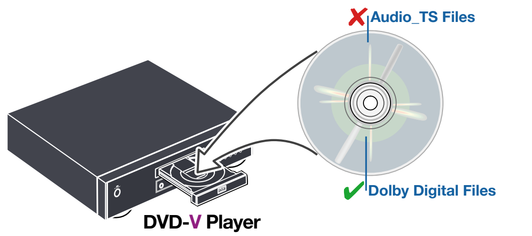
1. Scaleable PCM
2. Automatic downmixing
3. Optional surround audio formats
4. Still and full-motion video capabilities

Let's get the minor things out of the way first.

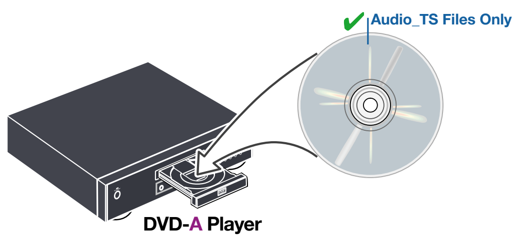
Automatic downmixing (also called "Smart Contents") uses a set of instructions "buried" in the multi-channel information to allow a DVD-A player to convert that data into a two-channel mix. Why use Smart Contents rather than put a separate two-channel mix on the disc? For one, there might not be room in the "audio-only" section of the disc to do so. And there might not be enough time – i.e., money – either. (Constructing a second "stereo only" two-channel mix means more time in the studio.) For these reasons, an artist or producer may code "fold-down" instructions into the multi-channel data that your player uses to construct a two-channel signal.



A DVD video-only player will NOT play the "AUDIO_TS" files on a DVD-A disc.



A DVD video-only player WILL play the optional Dolby Digital files on a DVD-A disc. However these are NOT the high definition "AUDIO_TS" files you may want to hear.



Only DVD-Audio player will automatically default to the high definition "AUDIO_TS" files you want to hear.

The ***optional surround formats*** include MPEG audio (both MPEG-1 and MPEG-2 varieties), DTS, and SDDS. And there's room on the disc for future formats we haven't even heard of yet.

Video support capabilities include:

1) a navigation menu very similar to those you've gotten used to (or incredibly frustrated by!) on DVD-V discs. Even though simplified (i.e., non-TV) navigation is available, it's optional and that means you'll probably have to turn on your TV to see what's on a DVD-A disc.
2) up to 16 still frame video images per audio track for liner notes, pictures, a libretto, maybe even the notes George Harrison scribbled during an early Beatles recording session!
3) limited full-motion video, too!

One other item may be of interest: *Regional Coding* will not be used on DVD-A discs. So you can buy a DVD-A disc anywhere and play it back anywhere.

With those items out the way, let's look at the more complicated issues.

Scaleable PCM

This simply means that a DVD-A disc may contain digital data in a variety of PCM structures determined by, among other things, artist or producer requests and the "bit budget" dictated by data density and disc storage capability.

First, content providers (artists, producers, record companies, etc.) can choose the bit length (16, 20, or 24 bit words) and the sampling frequency.

As you'll soon see, sampling frequencies are grouped into two families, "consumer" (44.1, 88.2, and 176.4 kHz) and professional (48, 96, and 192 kHz).

Notice that the sampling frequencies on each branch are mathematically related to each other. Everything on the consumer side is a multiple of the 44.1 kHz Red Book CD standard. The professional side is based on the 48 kHz sampling frequency first popularized by DAT (Digital Audio Tape).

Here are the guidelines artists or producers have to follow when choosing the data structure best suited to a particular project.

DVD-A allows up to 6 separate channels *UNLESS* the artist chooses a 176.4 or 192 kHz sampling frequency. These two "high-resolution" modes are limited to *two channels only*. Why? It's that "bit budget" again. These high sampling rates generate so much data that even a DVD can't hold six channels' worth!

But if all six channels on a DVD-A disc weren't equally important – as is often the case when the "surround" channels carry mostly ambient information – wouldn't it make sense to store that less important information in a less data-intensive form? Of course it would.

That's why DVD-A lets us choose different data structures for different channels when that's needed. And that's what we mean by "scaleability."

Is this a bit confusing? Yes, it is. But here's some information that'll help to clear the air.

Channel Groups

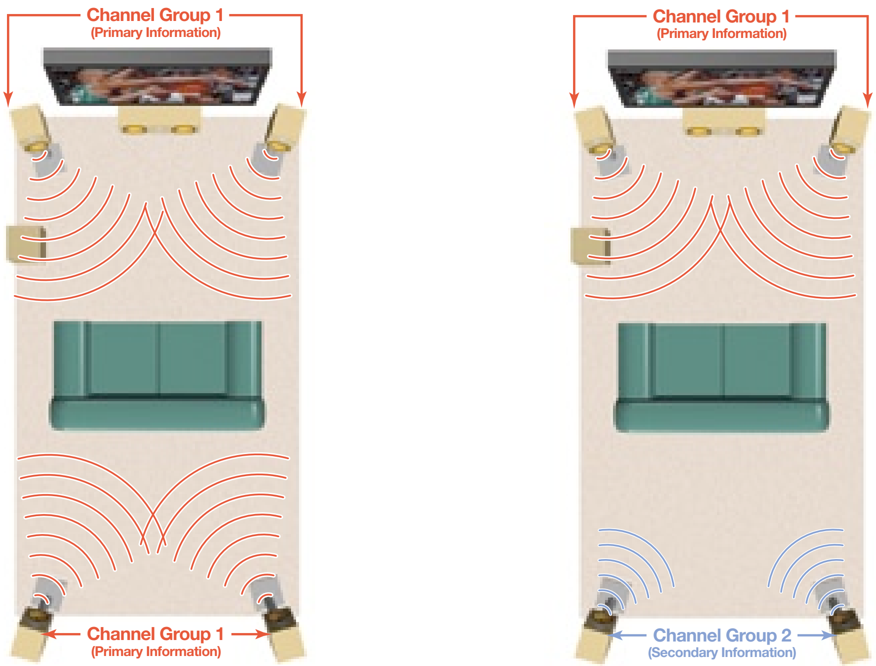
The DVD-A spec distinguishes between *primary* and *secondary* information.

Those channels carrying *critical data* are arbitrarily assigned to *Channel Group 1*.

Those with *less important information* are categorized as *Channel Group 2*.

These categories change depending on the material you're placing on a DVD-A disc. For example, a Left Surround channel may find itself in Channel Group 1 when carrying primary musical information or in Channel Group 2 when used for ambience reproduction.

Here's how this works: Category 2 channels can not – under *any* circumstances – have a longer bit structure or higher sampling frequency than Group 1 channels. In other words, Group 2 channels never convey more information than Group 1 channels. Never.



The same speaker falls into "Channel Group 1" or "Channel Group 2" depending on how important its information is.

Here's a table of the possible combinations. You'll notice that Group 1 channels are shown in unshaded boxes while Group 2 channels are in shaded boxes. You'll see why soon.

Channel Group 1	**Channel Group 2**	**Comment**
24 bit 192 kHz	None permitted	This format for "high resolution" monaural or stereo use only
24 bit 176.4 kHz	"	"
24 bit 96 kHz	24, 20, or 16 bit 96 or 48 kHz	Group 2 resolution can be the same but not greater than Group 1
20 bit 96 kHz	20 or 16 bit 96 or 48 kHz	"
16 bit 96 kHz	16 bit only 96 or 48 kHz	"
24 bit 88.2 kHz	24, 20, or 16 bit 88.2 or 44.1 kHz	"
20 bit 88.2 kHz	20 or 16 bit 88.2 or 44.1 kHz	"
16 bit 88.2 kHz	16 bit only 88.2 or 44.1 kHz	"
24 bit 48 kHz	24, 20, or 16 bit 48 kHz only	"
20 bit 48 kHz	20 or 16 bit 48 kHz only	"
16 bit 48 kHz	16 bit only 48 kHz only	"
24 bit 44.1 kHz	24, 20, or 16 bit 44.1 kHz only	"
20 bit 44.1 kHz	20 or 16 bit 44.1 kHz only	"
16 bit 44.1 kHz	16 bit only 44.1 kHz only	"

When Only the Right Combination Will Do

Now that you understand Group 1 and Group 2 channels, we're going to take the next step and show you all the combinations permitted by the DVD-A spec.

As you look over the next table, notice that we've used the same convention as in the previous table:

Group 1 channels are in clear boxes.

Group 2 channels appear as shaded boxes.

Unused channels in each combination are blacked out.

Obviously, some of these combinations will be far more common than others. For example, you'll probably see far more high resolution music-only discs in stereo (#2) than mono (#1). But all of them are numbered for your convenience.

Channel Combinations	Left	Center	Right	Left Surround	Right Surround	LFE
Sound Quality Priority: Mono and Stereo						
1 Mono						
2 Stereo						
3 Stereo with Mono Surround				(Mono)		
4 Stereo with Stereo Surround						
5 Stereo with LFE						
6 Stereo with Mono Surround and LFE				(Mono)		
7 Stereo with Stereo Surround and LFE						
8 Stereo and Center						
9 Stereo with Center and Mono Surround				(Mono)		
10 5 channel						
11 Stereo with Center and LFE						
12 Stereo with Center, Mono Surround, and LFE				(Mono)		
13 Stereo emphasis but with full 5.1 output						
Sound Quality Priority: Front Channel Speakers (L,C,R)						
14 Stereo with Center and Mono Surround				(Mono)		
15 Stereo with Center and Stereo Surround						
16 Stereo with Center and LFE						
17 Stereo with Center, Mono Surround, and LFE				(Mono)		
18 Front emphasis but with full 5.1 output.						
Sound Quality Priority: Corner Speakers (L, R, L Sur., R Sur.)						
19 Stereo with Stereo Surround and LFE						
20 Stereo with Center and Stereo Surround						
21 Corner speaker emphasis but with full 5.1 output.						

As you can see, combinations 1 through 13 are geared to two-channel (stereo) reproduction even though some of them allow surround, center, or LFE information, too. Numbers 14 through 18 cover home-theater applications while the remainder (19 through 21) handle the needs of the multi-channel music community.

Incidentally, if you want an objective look at performance potential, see the Sidebar entitled *Digital Performance in the Real World.*

Playing Time

Even DVD-A's enormous storage density doesn't allow willy-nilly use of space or totally unlimited playing time. This chart shows some of the limitations.

Audio Format	***Digital Data Structure***	***Playback time per side (4.7" disc)***	
		Single layer	***Dual layer***
Stereo	24 bit/48 kHz	>250 minutes	>460 minutes
Stereo	24 bit/192 kHz	>60 minutes	>115 minutes
Multi-channel	16 bit/96 kHz (x 6)	>60 minutes	>115 minutes
Multi-channel	20 bit/96 kHz (x 5)	>60 minutes	>110 minutes
Stereo **and** **Multi-channel**	24 bit/96 kHz (x 2) + 24 bit/96 kHz (x 3) 24 bit/48 kHz (x 2)	>40 minutes each	>75 minutes each

Enter Compression

When you take a look at the last block (24/96 stereo plus 5 channels of 24-bit multi-channel), you'll notice that there isn't much playing time available. Given that this might just be a popular format, the short playing time could be a real handicap.

That's what prompted the desire for some sort of data compression – or a way to cram more playing time on a disc – without compromising playback quality. The answer was a particular type of compression – *lossless.*

That's why MLP, or Meridian Lossless Packing, became part of the DVD-A spec. Developed by Michael Gerzon and Bob Stuart, this lossless compression is very effective in condensing data – in effect, getting more of it onto a disc so the playing time could be substantially extended.

MLP can result in up to a 50% space saving (in other words, a 2:1 compression ratio). And it's variable so it can be matched effectively to whatever type of music you're trying to compress. (Generally, simpler music can be compressed more.)

How effective is it? Here's your answer:

Format	**Uncompressed Playing Time**	**MLP-compressed Playing Time**
24 bit/192 kHz (x 2)	64 minutes	120-140 minutes (variable)
24 bit/96 kHz (x 6)	43 minutes	64-135 minutes (variable)

You should know that MLP decoding capability is *required* for all DVD-A players but is *optional* for content providers – not all DVD-A discs will make use of it.

LET'S GET PRACTICAL

OK, now that you know all the ins and outs of DVD-A, what will you need to enjoy it?

That's simple – here's the list:

1) A DVD-A player. Remember that all DVD-A players will be quite able to handle CDs. If a customer's old CD player is on its last legs, a DVD-A player may not be so expensive after all. And all DVD-A players will play DVD-V discs quite happily also.
2) Some DVD-A discs.
3) A 5.1 channel surround sound system. If you're a Home Theater 2.0 holdout (still happy with stereo sound, even from surround-encoded video sources), don't forget that DVD-A players and discs will work hard to bring you a stereo signal you'll be happy with.
4) A TV of some sort to help you find everything on a DVD-A disc and to watch whatever video information the creative folks included on the disc.

Some Considerations

Here's a short list of points about DVD-A you may want to think about. Feel free to add other items yourself.

The "better sound" potential will appeal to many, even when the differences may not be immediately apparent. That's a solid PLUS.

5.1 channel sound capability. If you enjoy being surrounded by music, this might be the real ticket. Once there's enough software out there, you'll have onstage seats for the performance of your dreams (or your nightmares if you prefer stereo's "up there in front" presentation). PLUS or MINUS.

A TV. If you're a died in the wool "turn the lights low and just listen" type, DVD-A's menu system will be a major annoyance. If you're more in tune with today's video-based entertainment, this won't bother you at all. And a TV will give you the window you need to see whatever visual information the artist and producer thought you'd like to see. PLUS.

DVD-A players, especially early ones, are comparatively slow in recognizing a disc. If you're used to the almost negligible delay between putting a CD in a CD player, hitting "Play," and hearing music, get ready for some l-o-o-n-n-n-g response times. There's a lot of digital "handshaking" and "initializing" before you'll hear the music but newer players will be quicker than older ones in this regard. (This also applies to earlier SACD players, too.) A minor (perhaps *very* minor) MINUS.

Software availability (at least for the next two or three years) compared to CD will be sparse. Any new format simply won't give you the variety of choices that CD, for example, gives you right now. But what else can you expect – remember that we've had CDs around for almost 2 decades! It'll take DVD-A, or any other new format, many years to catch up. A temporary MINUS.

Software confusion. Remember that DVD-V machines won't play the DVD-Audio tracks on a DVD-A disc even though they can play the Dolby Digital version of the same music – that is, *IF* the creative folks decided to include that Dolby Digital version in the first place! A possible MINUS only if you *don't* have a DVD-A player.

DVD-A'S FUTURE . . . ?

"So where is DVD-A going?" you might ask. Let's dust off the crystal ball and take a look.

Obviously, surround sound capability is a big plus as we move (some of us *very* reluctantly to be sure!) into this new aesthetic sensibility. That's something just totally beyond a CD's capability (except for the very few Dolby Digital or DTS-encoded "CDs"). Score a big one here for DVD-A.

Better sound quality, even for stereo sources? Frankly, that's a questionable virtue as far as the general public is concerned as a properly recorded CD is amazingly good to begin with. Will most people hear any difference between a conventional CD and a 24/192 DVD-A made from the same master source? We suspect not. On the other hand, the general public isn't a group of audiophiles.

Extended playing times? At first glance, this appears to be a very attractive benefit for DVD-A until you remember that this doesn't really have much to do with technology itself. This is really a matter of royalties. Just how much material will a studio want to cram onto one disc in the first place? And how much will you pay for it? We don't think you'll be seeing many 3+ hour DVD-As.

Video supplements? This is a very personal issue. We think this is a potential winner. For those who aren't interested, don't hook up the TV (and miss all those menu choices, right?).

Remember that DVD-A doesn't offer any convenience advantages over CD. Both have quick track access, remote control, etc. On the other hand, the format doesn't take any away (unless you're a speed freak when it comes to hearing that first cut!).

The key to DVD-A's success is software. If the ever-more-paranoid music business wakes up to the fact that *anything* can be hacked and opens the spigot for widescale DVD-A releases, we just might have a commercial monster on our hands. And a superb-sounding one at that!

PREVIEW

Like DVD-A, SACD is a high-density digital audio format based on DVD's large storage capabilities. Developed by Sony and Philips, SACD is aimed at sound-conscious consumers (as is DVD-A) but arrived on the scene first. SACD has already earned a reputation for excellent sound quality. Although first-generation players and discs were stereo-only, the format always provided for multi-channel capability.

Will SACD maintain its initial position? What are its strengths? And, perhaps even more importantly, what are its weaknesses?

You'll find out here.

Technically, SACD (the disc format) is based on a digital data structure called DSD (Direct Stream Digital) that is very different from DVD-A's more common PCM (multi-bit) structure. (For more information on this topic, see the Sidebar entitled *Multi-bit vs. Single-bit.*)

WHERE DID SACD COME FROM?

Some cynics say that SACD grew from Sony/Philips' realization that their CD patents were about to expire and, with them, royalties both companies enjoyed. Although there's *some* truth to this, the fact is that Sony (owner of Sony Music, Columbia, and other record labels) and Philips (then the owner of Polygram) knew they had to come up with a better mousetrap than the CD if they were going to archive the vast library of deteriorating analog master tapes in their libraries.

They developed DSD (Direct Stream Digital), a single-bit format created to preserve the musical nuances contained on those old masters.

Even though single-bit processing had, to their ears at least, some significant sonic advantages, they didn't ignore practical items like music replication and distribution either. So DSD had multiple goals: preservation, professional acceptance, and, finally, a place as a possible new consumer format.

But, as we've already explained, Sony and Philips ran into a major roadblock when they presented their development to the DVD Forum's WG4 (Working Group 4, the group setting the standards for all DVD audio formats). Suffice it to say that corporate rivalries carried the day and, after being firmly rebuffed, Sony and Philips decided to market this "maverick" format on their own.

WHO WANTS IT?

In some ways, an SACD disc is closely related to the CD. First of all, it is an *audio* format with admittedly fewer video attributes than DVD-A. For some, that may be bad news. But the other side of the coin is that you can navigate an SACD disc very well indeed without having to turn on your monitor, an attribute not always matched by DVD-A. For some "music-is-the-reason-I-exist" types, this might make SACD very attractive.

THE SACD DISC

Physically, SACD discs are all one sided. (There is some talk of dual side SACDs but nothing has appeared so far.) That one side may contain:

1) one high-density DSD layer
2) two high-density DSD layers
3) one high-density DSD layer and one CD-format layer.

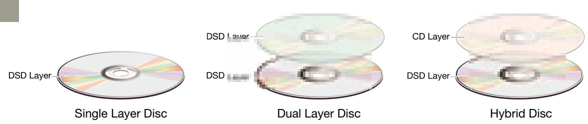
These DSD+CD dual layer discs (the third type) are called *hybrid* discs. They're intriguing because they're backwards-compatible, i.e., a conventional CD player will automatically read the CD layer while an SACD player can read either layer (but generally defaults to DSD data). You'll need an SACD player to read DSD-only (non-hybrid) discs regardless of whether they're single or dual layer.

A Quick Comparison

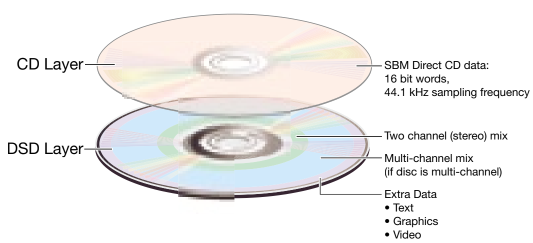
Before we look closely at an SACD disc, here's a quick comparison between CD and SACD that you might find interesting. Notice that we've put the performance specs first so you can get a better idea of the differences you may hear with SACD.

	CD	**SACD**	**What The Difference Means**
Performance Specification			
Coding System	16 bit linear PCM	1 bit DSD	16 bit PCM is somewhat limited but still very good. DSD is theoretically better but practical things like recording and mastering quality will make more of a difference in what you hear.
Sampling Frequency	44.1 kHz	2.8224 MHz	This is just a natural consequence of the differences between CD's multi-bit PCM structure and DSD's one-bit format. Don't let anyone tell you that the sampling frequency itself means better sound.

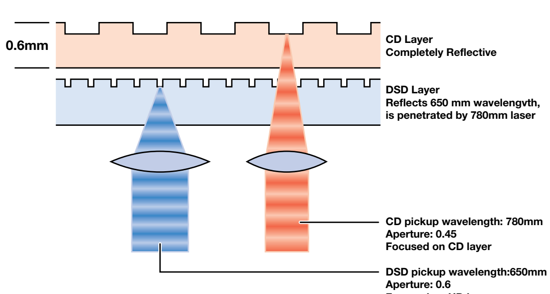
Frequency Response	DC (or 0 Hz) - 20 kHz	DC - 100 kHz	A practical advantage for SACD here as we respond to frequencies higher than 20 kHz even if we don't "hear" them as such.
Dynamic Range	96 dB	greater than 120 dB	Of almost no practical advantage for *home* use.
Maximum Playing Time	74+ minutes	110+ minutes (2 channel) 80+ minutes (multi-channel)	SACD is certainly more convenient for long-term playback but disc changers take away a lot of that advantage.
Physical Specifications			
Disc Diameter	12 cm (4.7")	12 cm (4.7")	Same diameter.
Disc Thickness	1.2 mm	1.2 mm	Same thickness
Playback Sides	1	1	Same number of sides
Laser Wavelength	780 nm	650 nm	SACD's shorter wavelength laser means that it can focus on smaller "pits" on the disc. Note that SACD players need a dual wavelength laser to read both SACD and conventional CD discs.
Pit length	0.83 µm	0.40 µm	SACD's smaller pits mean that more of 'em can be crowded onto a disc. The result? More data.
Track pitch	1.6 µm	0.74 µm	SACD's track "spiral" is also tighter so a disc holds more data.
Data Capacity	650 megabytes	4.7 gigabytes (DSD layer)	MUCH more data capacity for an SACD disc.
Additional Capabilities			
Text	Yes	Yes	Comparatively few CD players have "CD Text" capability. Very few discs carry "CD Text" information in the first place.
Graphics	No	Yes	A clear advantage for SACD – *IF* you want graphics.
Video	No	Yes	Ditto.



The SACD standard allows three kinds of discs: a single-layer disc with DSD data only, a dual-layer disc with both layers carrying only DSD data, and a "hybrid" disc with both a DSD layer and a conventional CD layer.



From the inside out, a DSD layer contains: (1) a master table of contents (TOC), (2) 2-channel (stereo) audio, (3) multi-channel audio, and (4) "extra" data.



The genius is in the backing. The surface above the CD (top) layer is completely reflective while the surface above the DSD layer is semi-transparent. When hit by the 650 nm "high density" laser used to read DSD data, it reflects the signal back to the optical system. But 780 nm "CD laser" passes right through it to the layer containing the CD data.

Bits, Bytes, Nuts, Bolts

Despite all the negative hue and cry about single-bit digital processing common in the late '80s when Panasonic's poorly named MASH processors hit the street, there's something very elegant about the idea.

DSD takes advantage of single-bit simplicity with a cleaner path from analog to digital and back again than PCM is capable of. The reasons for this are quite complex (even if the circuitry is simpler!) so we'll just ask you to take our word for it right now.

One Disc for Every Player?

Let's look more closely at one of SACD's most interesting aspects – the promise that one disc might play in a conventional CD player (and produce quite acceptable CD-quality sound) and yet reveal even more detail in an SACD player! As we've hinted at before, here are the disc types defined by the SACD standard.

The "extra" data can be text – song lyrics, liner notes – or even still images to complement the sound. (But note that the video imaging capability of an SACD disc is considerably less than you'll find on most DVD-A discs.)

Hybrid Appeal

The hybrid (DSD/CD) disc holds obvious appeal for many potential listeners. They can play the same disc in *any* CD or SACD player. A conventional CD player reads the PCM-format "CD layer" only. Although SACD players will default to the higher resolution DSD layer, a listener can choose the CD layer if desired!

1 + 1 = 3?

SACD's backers claim another advantage for the hybrid discs. In addition to the improvements offered by the DSD layer, they're also saying that the CD layer will yield better sound than you'll get from a conventional CD of the same music played on a conventional CD-only player. Sony and Philips attribute this to something called Super Bit Mapping Direct, an offshoot of Sony's earlier Super Bit Mapping (or SBM).

This technology is theoretically intriguing but we haven't had the opportunity to sit with two players (SACD and CD) with an SACD and CD made from identical masters so we can't vouch for its effectiveness in the real world.

Commerce Rears Its Ugly Head

Now that you understand the technology behind the hybrid disc, you're in a really good position to see the potential commercial advantages.

If record companies adopt the SACD format, so say its advocates, they won't have to produce two versions (CD and "high resolution") of the same music. And record stores won't have to inventory them either! That's a significant theoretical advantage for SACD.

Truth be told, there has been some talk from the DVD-A camp about hybrid discs too. But WG4 spokespeople have been very negative about manufacturing problems and associated costs so we don't think you'll see a dual-density DVD-A disc soon. But stayed tuned for further developments.

Distributing the Music . . .

In addition to the attractions inherent in single inventory, SACD and DSD appear to have an edge in getting music from the recording and mastering studios into your hands. This has to do with something called "downconversion" – or the ease with which a DSD-master can be converted to today's popular digital formats.

This is in stark contrast to DVD-A's "Chinese menu" of various PCM data structures. By now, you're aware that DVD-A's various sampling frequencies fall into two "families": consumer-oriented (based on the CD standard of 44.1 kHz) and professional (with a 48 kHz base). Within each family, converting from one sampling rate to another is easy. Want to go from 44.1 kHz to 176.4 kHz? Easy – just multiply by 4!

But going from one family to the other, as often happens when a studio master tape at 48 kHz is destined for release on CD at 44.1 kHz, is a mathematically complex story. Although today's DSP processors can *do* it, it is a difficult process.

One of DSD's theoretical advantages is that it allows easy downconversion to almost any of today's PCM formats. Once in DSD form, simple multiplication and division converts a musical or movie soundtrack event for commercial distribution. Audibly, this process is more benign than the inter-family PCM conversion problems we just outlined.

Are You Compressed?

Even though an SACD holds much more data than a conventional CD, DSD's greater data load really strains the available real estate. For that reason, the SACD format allows a form of data compression called Direct Stream Transfer or DST. Developed by Philips, this lossless 2:1 compression scheme guarantees bit-perfect reconstruction of the original signal and allows extended playback times. For example, DSD's high-density two channel stereo tracks can be as long as 110 minutes. Multi-channel segments can last as long as 81 minutes. (Notice that there are no quality trade-offs when you go from two channel to multi-channel as there are with DVD-A. Data density for both stereo and multi-channel versions of the music is the same. In other words, there's no "scaleability" involved.)

Are You Legitimate?

The Copy Police are patrolling SACD's neighborhood too. In response to the music industry's concerns about unauthorized copying and duplication, the digital outputs on early SACD machines will output only CD-quality audio. You won't be able to access the DSD data at all. In addition, they've added a physical watermark system that allows content providers to trace the source of discs suspected of being counterfeit.

Pit Signal Processing (or PSP, as it's called) modulates the width of a disc's pits. (Don't worry about the audio data itself – *that's* dictated by the length of the pit.) Circuitry in the player reads the watermark and compares it to a "key" buried in the data. PSP also creates a visible holographic-like pattern on the disc surface so it's easy to tell a legitimate disc from a counterfeit one.

LOOKING INTO THE FUTURE

Does SACD have the "legs" it needs to become a truly important part of the consumer electronic scene? Maybe. The technology itself is fascinating but, until recently, the format did not really enjoy the software backing it needed to insure success.

Most of the big record labels have backed DVD-A since the format was first announced. Although new title introductions have been slower than initially expected, that rate is increasing substantially now. How this will affect SACD is unclear right now. Although SACD gained most of its initial support from smaller audiophile-oriented labels like Audioquest, dmp, and Telarc, recent announcement from some of the "big boys" (Sony Music – of course!, EMI, Universal, Virgin, etc.) seem to make this "format skirmish" very interesting indeed.

SO WHERE DO WE STAND?

Before we go promoting either DVD-A or SACD at the expense of the other, let's look at some facts.

1) Both formats can deliver better potential sound quality than CD.
2) Both formats are capable of multi-channel sound, something that CD – except for the comparatively few Dolby Surround or DTS-encoded discs – simply can't deliver.
3) Both formats support video capability, another thing CDs can't offer.

That means both formats are better suited for today's multi-media entertainment needs than any we've had before.

That being said, there are differences. Each format has particular strengths.

DVD-A is the clear winner in video capability. Because the format is more closely related to DVD-V than is SACD, DVD-A discs can bring us closer to a musical event through full-motion video clips and a veritable barrage of still frame support. SACD can't match these features and if they're important to you, then DVD-A is preferable.

On the other hand, the "buzz" is that SACD has the edge in ultimate sound quality. But that may only be relevant if the original recording used this technology. If not, then it's a moot point. There isn't much of an argument that DSD's data structure is better for archival storage and data reformatting/distribution purposes. Although this last attribute may be of greater importance to the music industry than it is to you, it is something to be considered.

For those of you interested in yet one more table, here's a comparison between DVD-A and SACD.

	DVD-A	**SACD**
Backers	WG-4 of DVD Forum (40+ members)	Sony & Philips
Digital Data Structure	PCM (multi-bit)	DSD (single bit)
Bit length(s)	16, 20, & 24 bit	Not Applicable (see above)
Sampling Rate(s)	32, 44.1, 48, 88.2, 96, 192 kHz	2.8224 MHz
Disc Format	Single side/single layer *for now;* single side/dual layer and dual side, dual layer *possible.*	Single side/single layer Second layer — either DSD or Red Book (CD) — *optional*.
Number of Channels	Up to eight discrete channels (5.1 + stereo) per layer.	Up to eight discrete channels (5.1 + stereo) per DSD layer. Red Book layer (if used) limited to 2 channels.
Compression?	Yes. Lossless with 2:1 compression ratio. MLP (Meridian Lossless Packing) mandatory for hardware, optional for software.	Yes. Lossless with 2:1 compression ratio. DST (Direct Stream Transfer) mandatory for hardware, optional for software.
Play Time	Program/MLP dependent: 74 to 120 minutes of two channel 24/192 data per layer, up to 89 minutes of 5.1 channel 24/96 data per layer.	Also program/DST dependent. 110 minutes maximum per layer for 2 channel DSD, 81 minutes for multi-channel DSD per layer. Red Book layer (*if used*) limited to 80 minutes. .
Music Industry Support	Widespread.	Sony Music, dmp, Telarc, Water Lily, Audioquest, etc.
Studio Mastering Equipment	Available from several sources.	Also available but mainly from Sony subsidiary Sonoma Systems.
Player and Disc Availability	Many players available from wide variety of manufacturers. Disc title list growing significantly.	Stereo and multi-channel players and discs now widely available.

TECHNICAL APPENDIX

1) Digital Performance in the Real World

> ***Here's some information you should have seen before – you'll find a slightly different version under "Digital Performance Parameters" in Section 2. However it's important enough to read again so here it is!***

PCM digital signals (the format used for both CDs and DVD-A discs) are usually composed of groups of 16, 20, or 24 "bits" (0s or 1s) called "words." These words move through a digital system at a certain rate called the "sampling frequency."

Bits . . .

The number of bits in each digital "word" directly affects performance: More bits generally means higher resolution and better signal to noise performance. Here's a quick guide:

System bits	*Expressed As*	*Possible Quantization Levels* (2n where n = number of bits)	*Theoretical S/N* (-6 dB/bit)
16	2^{16}	65,536 [1]	-96 "
20	2^{20}	1,048,576 [2]	-120 "
24	2^{24}	16,777,216 [3]	-144 "

Notes:

[1] "Red Book" CD standard, [2] Early digital studio mixers, editors, etc.,

[3] Today's studio technology; also used for DVD-A

As you can see, a 24-bit signal is theoretically capable of resolving a musical signal 256 times better than a 16-bit signal. (Divide 16,777,216, the number of quantization levels available in a 24-bit system, by 65,536, the limit for a 16-bit system.) In addition, look at the differences in potential signal-to-noise performance. (But remember that theory does not always translate to the real world.

. . . and Samples

The next thing we need to consider is "sampling frequency" or the rate at which these multi-bit words zip through our digital audio system.

In the PCM digital world, the highest audio frequency a system can convey is *half* its sampling frequency. So, if your digital phone answering machine had a sampling frequency of, say, 3 kHz, the highest audio signal it could process would be 1.5 kHz. (And that's just fine for voice.) CDs, with a sampling frequency of 44.1 kHz, go up to 22.05 kHz.

Here's the same idea in chart form:

Common sampling frequencies	*Highest audio frequency* Theoretical/Practical
44.1 kHz	22.05 kHz/20 kHz
48 kHz	24 kHz/22 kHz
96 kHz	48 kHz/44 kHz
192 kHz	96 kHz/88 kHz
2.8224 MHz (SACD)	100 kHz/100 kHz [1]

[1] Remember that the SACD's high-density data structure is single-bit delta/sigma rather than the multi-bit PCM format specified for CDs and DVD-A discs.

> ***Note: There's a difference between a system's basic sampling frequency and something called an "oversampling" rate. Oversampling (we should call it "resampling," by the way) is a*** playback ***spec only. You'll notice that resampling rates are always even multiples of a system's basic sampling rate. (A CD player with "8x" resampling, for example, processes data at 352.8 kHz – exactly 8 times the CD-standard sampling frequency of 44.1 kHz.)***

Don't look for resampling or oversampling specs in DVD-A players. For a variety of reasons, the technique isn't important.

When Reality Intrudes

Sometimes these numbers are confusing as higher ones don't *always* guarantee better sound.

For example, a poorly designed 24-bit system probably won't convey the same level of musical information as a well-implemented 16-bit system. And high sampling rates (88.2 and above) do not necessarily mean better performance either. High levels of jitter, for example, may make a 96 kHz system less musical than one that operates at 48 kHz. .

Even high-density D/A converters are rarely capable of signal-to-noise performance better than –110 dB because of something called thermal noise. (If we could place our components in tanks of liquid helium or something equally cold, that would literally "freeze out" thermal noise. But that's not very practical, is it?)

Power supply noise is another real-world limiting factor. It's rarely better than –110 dB and it, too, impacts a digital system's low-level resolution.

Since we're talking about recorded events, we also have to consider the noise levels inherent at the very beginning of that chain – the microphone. With few exceptions, performance levels for this critical link simply do not exceed –90 dB.

That's why theoretical signal-to-noise and dynamic range specs for only the digital data structure itself don't tell the whole story.

2) Data Compression

Data compression is a complex topic. But don't get alarmed. We're going to make it simple.

In the digital world, there are two types of compression: **lossy** and **lossless**. They both serve the same purpose – to fit more data into (or onto) a space that isn't large enough to hold all of it under normal circumstances.

As you can probably tell from the name itself, **lossy compression** does this by simply discarding (or losing) some of the data. This isn't done arbitrarily, by the way. Instead, a computer looks at the input signal to determine what's essential and what isn't. It's guided by a program (called an "algorithm") based on our ability to hear certain kinds of sounds and, equally important, our *in*ability to hear others.

The computer looks for *redundant* or *irrelevant* data, sounds that are either needlessly repetitive or ones we can't hear because they've been masked by other sounds, and simply throws them away.

The result is less data that, naturally enough, takes up less space.

Of course, lossy compression can be fairly innocuous or really noticeable. It all depends on how well the algorithm "tracks" our hearing (some programs are much better than others) and how much compression the engineer cranks in during production (most algorithms are adjustable from "let's slim this down just a bit" to "let's squash that sucker flat.")

The problem with lossy compression is that once we've thrown away part (even an "inaudible" part) of the signal, it's gone. As in *GONE*. That makes lossy compression totally unsuitable for any kind of archival purpose where we want to preserve all the sonic complexities of the original. It also makes lossy compression a dubious choice for anything that aspires to "high fidelity" as, by its very definition, lossy compression is not faithful to the original.

Enter **lossless compression**, the second and – at least for our purposes – better choice. Lossless compression allows us to completely reconstruct the original signal after we've stored it or transmitted it. And this isn't just a closer approximation to the original by the way but a bit-for-bit reconstruction – you can't tell the reconstructed information from the original in any way.

Lossless compression works by "packing" the information more efficiently. It doesn't discriminate between "good" data and "unnecessary" data. It simply packs it all into as tight a bundle as it can so it can be transferred more efficiently.

The best example of lossless compression we can think of is the ubiquitous "WinZIP" utility almost all computer users use to transmit large files. A "zipped" spreadsheet is every bit (if you'll pardon the pun) as accurate as the unzipped original once you've uncompressed it. And a music file, whether a song or a symphony, is note-for-note, nuance-for-nuance identical to the original.

3) Multi-bit vs. Single bit

DVD-A uses a particular type of digital data called PCM (for Pulse Code Modulation). So do CDs. PCM's most salient feature is that it is a "multi-bit" format. That means every digital "word" (a 16, 20, or 24-bit entity) *completely* describes a sample's amplitude. Although accurate, PCM is data-intensive because each sample stands completely on its own. PCM doesn't let you say, for example, "This sample is exactly the same as the previous one, so just repeat it."

"Delta/sigma" processing, or "single bit" technology, is entirely different. Rather than use PCM data blocks, a high-speed, single bit datastream describes amplitude *differences* from sample to sample. ("This one is the same as the one before . . ." "This one is lower in amplitude . . ." "This one is higher . . ." etc.) So, delta/sigma processing is less data-intensive than PCM.

Here's another way to look at the differences:

In moving digital data from one place to another, multi-bit PCM uses 16, 20, or 24 cups for each digital word, depending on the bit structure of the system. Each cup is a different size but their sizes are all related to that of the first. For example, if the first cup (the digital word's Most Significant Bit or MSB) in a 16-bit system held eight ounces, the second would hold four, the third two, and so on down to the last cup (the Least Significant Bit or LSB), which would hold a mere .00024414 ounces!

These cups would be used in varying combinations to describe the amplitude of each sample. As you can imagine, the size ratio from cup to cup is critical. Any variation, however small, would make the whole process inaccurate. We usually refer to these inaccuracies as "linearity problems."

Delta/sigma processing, on the other hand, uses literally thousands of much smaller cups, each one exactly the same size as all the others. And these tiny cups move much more quickly as well. (That's why most delta/sigma "sampling rates" are as high as they are.) The whole question of linearity, so important in multi-bit PCM processing, simply doesn't exist..

Although delta/sigma is not totally free of problems, they tend to be smaller in comparison to PCM's linearity challenges.

PREVIEW

Fact: No active component (repeat – NO ACTIVE COMPONENT) in a system contributes more to the sound you hear than loudspeakers.

Amplifiers don't. Surround processors don't. Cables certainly don't. Phono cartridges come close but that's because they, like loudspeakers, are transducers that convert one form of energy into another.

For most people (i.e., those without a turntable), the only real competition speakers have in shaping our aural environment are the rooms into which we place them. If you press the issue, some acousticians will tell you that speakers and rooms really can't be considered separately, that the speaker/room interface is so critical that it is a component in and of itself.

But rooms are admittedly difficult to deal with. They're just there. But if we understand more about speakers, we can begin to solve the "good sound" puzzle more efficiently.

In this Section, we'll describe different speaker types and their performance characteristics. We'll make a few suggestions, dispel some noxious myths, and try to restore some sanity to an area that is sometimes curiously devoid of that quality. After you've read this, you'll be able to better understand why speakers are as important as they are.

One thing you should remember, however, is that speaker evaluation and selection is as much a subjective exercise as an intelligent evaluation of "proper" design parameters. In the final analysis, ears are the only arbiter. And that's just as it should be.

CHOICES

We usually group loudspeakers in a variety of ways: driver type, enclosure type, radiation pattern, etc. A total count of the possible combinations and permutations would tax a Cray supercomputer but we'll try to simplify things here.

We'll start with:

- an overview of what a loudspeaker is
- what parts you'll find in a typical example
- how speakers work to give you the sound you want

Drivers, the active elements in any speaker, are most often known by the range of frequencies they are intended to reproduce.

- *Woofers*, for example, handle low frequencies.
- *Midrange* drivers handle the critical middle range of audible frequencies. (Midrange drivers used to be called "squawkers" but no one uses that term anymore. Not hard to figure out why, is it?)
- *Tweeters* reproduce high frequencies

In addition to these generic classifications, you'll see an occasional reference to "subwoofers" and "supertweeters." *Subwoofers* handle (or are *supposed* to handle) frequencies below the range of conventional woofers. *Supertweeters* extend a speaker's ability to reproduce highs to a point where they may really irritate your dog.

On occasion, you'll see the term "full range driver." Be careful here. The physics of sound reproduction effectively prohibits a single driver from reproducing the entire audible frequency range. Either a "full range driver" really *isn't* or the manufacturer is using some heavy-handed electronic editing (read "equalization") to temporarily rewrite the laws of physics.

WHAT'S IN THAT BOX?

Most loudspeakers today consist of three important elements:

- the driver or drivers that actually produce sound
- a crossover network that makes sure that the drivers get that portion of the audio signal they're best able to reproduce.
- the enclosure the drivers are housed in

ANTI-SOCIAL AIR

Air molecules are anti-social. That's right, they don't like each other. When you cram them together, they pull apart. In fact, their anti-social tendencies are so strong that they over-react, pulling so far apart that they soon realize their mistake and opt to resume a relationship, rather distant to be sure, with their neighbors.

Of course, scientists would describe things a bit differently. They'd start with a definition of "equilibrium," that at-rest condition in which air molecules are at optimum distance from each other.

But, of course, nature is never at rest and this equilibrium condition is disturbed when something moves, expands, explodes, . . . or collapses. It's the disturbance that creates a sound wave. Here's how:

If you clap your hands, you're forcing air molecules closer together. This is called "compression" and, as we've already said, air molecules don't like that condition at all. So, in seeking equilibrium, they pull back from each other. This is called "rarefaction."

We define a sound wave as a series of these compressions and rarefactions. More specifically, air molecules that move from a compressed to a rarefied state more often than 20 times per second (or 20 Hz) but less that 20,000 times a second (20 kHz) create audible disturbances we hear as sound.

One thing to note about sound – the air molecules that impact your ear drums are NOT the same ones that responded to the initial disturbance – individual molecules remain pretty much where they were initially. Rather, it's the series of compressions and rarefactions that move out from the initial disturbance. Of *course, each molecule is subject to some back and forth motion (that's what transmits the wave after all) but no molecule is pushed along with the wave.*

If you watch a piece of driftwood or a cork bobbing in the surf, it won't move closer or further away at anywhere near the rate you might expect from the speed of the waves themselves. Yes, the cork or driftwood will rise and fall but it will remain pretty much in the same place otherwise.

When you think of the cork or driftwood as one air molecule and the surf as a sound wave, it's a lot easier to understand how air molecules respond, isn't it?

DRIVERS

Dynamic Drivers

The vast majority of speakers sold today use so-called **dynamic** or **moving coil** drivers to sequentially compress and then rarefy air molecules; in other words, to create sound waves that we hear.

Most dynamic drivers are visible as soon as you remove the speaker's grill cloth. They're generally mounted on the speaker's baffle, the enclosure's front panel.

When we look at a dynamic driver from the outside of an enclosure, we're looking at a part sometimes called the **diaphragm**, or the moving portion that actually creates sound.

Because they're comparatively small, particularly in comparison to planar drivers (wait, they're coming next!), dynamic drivers, especially woofers, must move in and out a fairly long distance to get those air molecules in motion. We use the term **excursion** to describe just how far a dynamic driver's cone or dome will move compared to its quiescent, or resting, state.

Dynamic Pros & Cons

There are so many variables in dynamic drivers that we're hard pressed to point out anything specific. Instead, we'll just say that dynamic drivers are ubiquitous – they're used everywhere, from down-and-dirty "punish your ears" speakers to some of the most expensive and revelatory products you'll ever listen to.

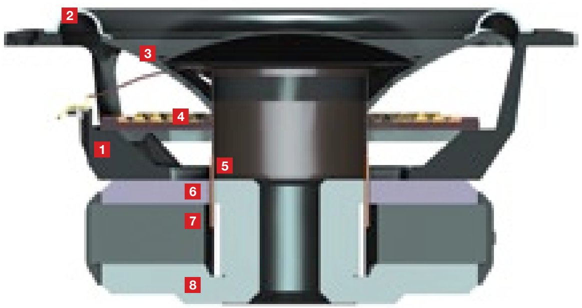
THE INSIDE VIEW

When you look at a conventional dynamic driver, you see something that's surprisingly complex.

Here's a view you probably haven't seen before.

The ID tags say:

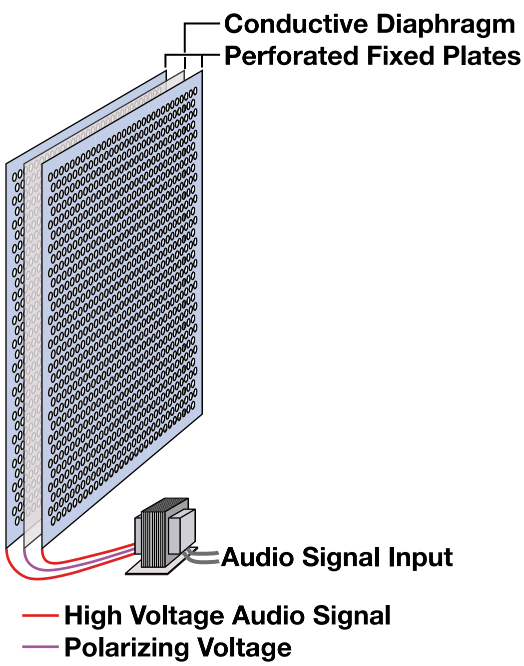
1) *Frame: This holds everything else together. Although plastic and carbon fiber are sometimes used, most frames are either cast or stamped metal. We prefer cast frames as they're usually stronger and more dimensionally stable.*
2) *Front surround: An important part of a dynamic driver's "suspension," this*



1) Frame
2) Front surround (corrugated circle between frame and front of cone)
3) Cone
4) Spider (support at rear of cone)
5) Voice coil on "former"
6) Front plate of magnet structure
7) Permanent magnet with circular gap clearly visible
8) Rear plate of magnet structure

centers the open end of the cone or dome in the frame. Surrounds come in a variety of shapes ("half roll" or "corrugated" are the two most common) and materials.

3) *Cone: The moving surface of a driver is sometimes called the diaphragm and a generally accepted design goal is to make it as stiff and inflexible as possible so that it operates as a true piston. Cones and domes (common diaphragm shapes) are made from a wide variety of materials from seaweed pulp to titanium, each material supported by a seemingly convincing story. Trust your ears.*
4) *Spider: Usually found only on woofers or large midrange drivers, this serves the same purpose as the front surround: it centers the cone, this time at the rear.*
5) *Voice coil on "former": This is the moving part of the driver's motor. The coil may be more complex than it seems – "edge" wrapping and multi-layer construction are variables you'll find in more exotic designs.*
6) *Front plate of magnet structure: This helps focus the magnetic field on the "gap" of the magnet.*
7) *Permanent magnet: This is the passive (i.e., non-moving) part of the motor. Magnets can be made of a variety of materials: ceramic, Alnico, rare-earth (neodymium), and others. As with cone materials, each magnet type has its supporters and the supporters all have their stories.*
8) *Rear plate of magnet assembly: Does just what the front plate does – helps to focus the magnetic field.*



Electrostatic speakers are known for their "speed" and detail. This quality results from the fact that the moving diaphragm is very light and is evenly driven over its entire surface.

Planar Drivers

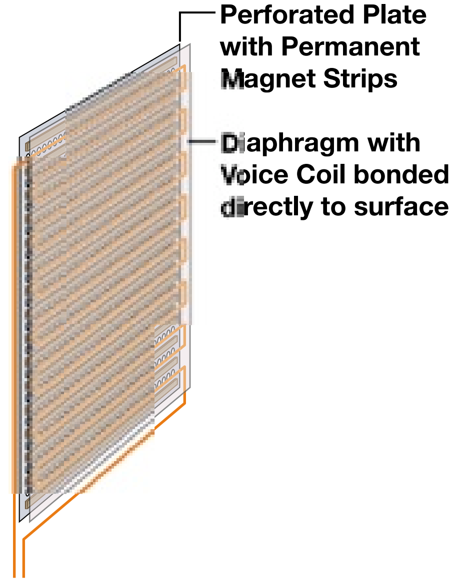
Now that we've explained dynamic drivers, let's take a look at two more types loosely grouped under the generic term **planar** drivers.

In general, planar drivers don't use the comparatively small surface area typical of a dynamic driver to move air. Instead, they depend on a much larger radiating surface. And they're dipolar speakers to boot – they move air equally well in front of them and behind them but these front and rear waves are out of phase with each other.

Electrostatic drivers: These are substantially different from dynamic drivers even though they perform the same function (i.e., get air molecules to move).

Rather than the "voice coil in a magnetic gap" motor, an electrostat (the short name) sports a large, ultra-thin sheet of material (think "Saran Wrap") suspended between two perforated plates. When the plates are charged by something called a "polarizing voltage" and then excited by the audio signal from your amplifier, the thin membrane moves back and forth in response to the audio information. This movement, in turn, displaces air molecules to create sound waves. Electrostats are dipolar radiators

Electromagnetic drivers: These use the same large Saran Wrap-like sheet as electrostats but there's a crucial difference – an elongated "voice coil" is printed on the membrane which is then suspended in a field made up of a series of small magnetic assemblies. The membrane moves back and forth as dictated by the audio signal passing through the "voice coil," thereby producing sound. So-called "ribbon" drivers (used primarily to reproduce high frequencies) are electromagnetic in nature.



Like electrostats, electromagnetic drivers respond very quickly to a signal and can be extraordinarily transparent. Like all dipoles, however, they are sometimes very difficult to position properly in a room.

Planar Pros & Cons
Planar drivers are, pardon the pun, polarizing – some folks love 'em, some don't have any time for 'em at all.

The good news is that planar drivers are usually very quick to respond to an input signal. That's because they have comparatively little mass. (Mass = inertia, etc.) Planar partisans point to the detail ("transparency") they deliver. Planar critics point to their usually large size, compromised bass response, and real difficulty in placing them properly in a room.

Horns
Are "horns" drivers in the strict sense or an enclosure type? It all depends.

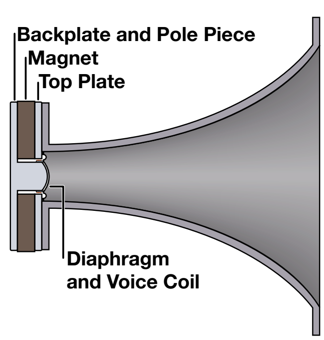
You've heard references to both "horn drivers" and "horn-loaded enclosures." There's some truth to both phrases so we'll start our discussion with horns as drivers.

In its simplest form, a horn is simply a flared pathway for sound. Megaphones are horns. Early Victrolas used acoustic horns to magnify the sound from a disk's surface.

An early "acoustic horn" record phonograph

Today, horns are used far more often in commercial and professional circles than in consumer products.

A typical horn driver uses a highly specialized dynamic element (usually called a "compression driver") to project the sound into the "mouth" of the horn. Once the sound wave begins traveling away from the throat, the horn's flared sides add to the efficiency as well as help control the sound wave's dispersion.



Horn Pros and Cons
Horns are very efficient and convert a lot more electrical energy from an amp into sound than almost any dynamic or planar driver. And horns are usually very directional – they project sound in a more precise way than other driver types do – that's why they're so popular for PA and sound reinforcement systems.

A horn's biggest downfall is just that – it must be big indeed to handle bass frequencies adequately. As homes today seem to be getting smaller, horns – even though they offer important performance advantages – have mostly fallen out of favor for domestic use.

CROSSOVERS

As we've just seen, the reason we have woofers, midranges, and tweeters is because each driver type is optimized to reproduce a particular range of frequencies.

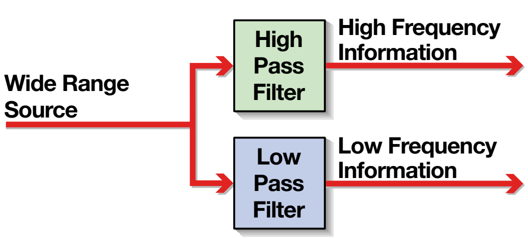
But how do low bass organ tones, for example, get to the woofer? And how do cymbal clashes get to the tweeter?

That's simple – the crossover tells 'em!

A crossover is an electrical circuit that divides a full-range signal (from an amplifier, for example) into smaller "bands" of frequencies. Technically speaking, a crossover is usually composed of at least two filters, a "low-pass" and a "high-pass" filter. Don't let the names confuse you.

A high-pass filter "passes" only high frequencies and blocks (or *attenuates*) low frequencies from appearing at its output. That's why it is used to condition a signal before the signal gets to the tweeter. If you routed a bass-heavy signal to a tweeter, it would probably burn out the tweeter's relatively small voice coil.

Similarly, a low-pass filter allows only low frequencies to pass through. It blocks, or *attenuates*, higher frequencies. Although there's little chance that a high-frequency signal would actually damage a woofer's voice coil, there's no reason to send a woofer something it can't reproduce anyway.



Block diagram of a simple two-way crossover. Notice that the wide range input signal goes to both the high-pass and low-pass filters at the same time. From there, only high frequencies go to the tweeter and only lows go to the woofer.

How Many "Ways" Are There?

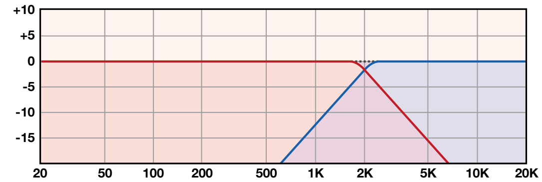
To understand this better, look at the block diagram of a simple "two-way" crossover – one with just one low-pass and one high-pass section.

> Note: From a functional view, this two-way crossover (with low-pass and high-pass filters) is one of the simpler crossovers imaginable. But it is not the simplest. That honor belongs to a crossover design you might call one-way in that it uses only a capacitor to block low frequencies from the tweeter.

Another way to look at a crossover is to imagine the theoretical "perfectly flat" frequency response curve we want to see from our speakers. As you know now, that response is really the acoustical combination of outputs from two different drivers, the woofer and the tweeter.

If we graph those outputs, we would see something like this:

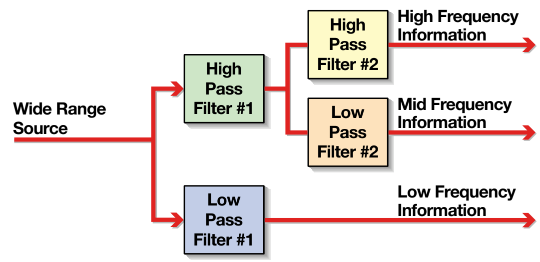
Look at the block diagram of a three-way crossover. Of course, this is a bit more complicated than a two-way crossover. But it does follow the same logic and execution. Four-way and five-way crossovers are really more of the same.



A two-way crossover routes low- and high-frequency signal components to the appropriate drivers which, in turn, produce sound.

Another Means To A "Way"

So far, we've used the word "way" to look at crossover design. Of course, that word can have another meaning. You've undoubtedly heard the expression "two-way speaker." (Just a few paragraphs ago, as a matter of fact!)

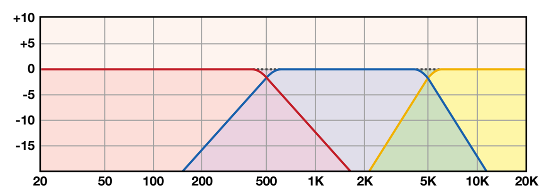


Block diagram of a three-way crossover. Note that the output of the first high-pass filter feeds a second filter set consisting of low-pass and high-pass sections.The outputs of each of these feed the midrange and tweeter respectively.

In its simplest usage, the term "two-way speaker" means one that has a woofer to reproduce lows and a tweeter to reproduce highs. In reality, two-way speakers use both the woofer and tweeter to reproduce the midrange too. After all, these designs attempt to cover the entire audible frequency range with just two drivers.

So, do all two-way speakers have a two-way crossover? Not necessarily! In fact, we've already given you an example of a "one-way crossover" that just blocks potentially destructive low frequencies from burning out a tweeter's relatively delicate voice coil.

And there are other kinds of possible "mismatches." One low-pass filter could, for example, feed two different woofers, each with a different cone material, for example, and each tuned differently – or in a different subsection of an enclosure – to cover a different portion of the low-frequency spectrum. In this case, you could have a two-way crossover in a "three-way" loudspeaker system.



This three-way crossover routes low-, mid-, and high-frequency signal components to the appropriate drivers which, in turn, produce sound. As in a two-way crossover, the "transition bands" where two drivers produce the same frequencies are crucial and engineers spend a great deal of time making sure that crossover characteristics and driver parameters match precisely.

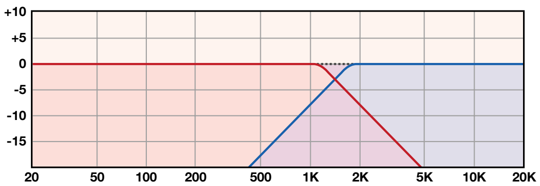
Filter Types, Slopes, and Corner Frequencies

The filter sections that make up a crossover are more complex than just "high pass" or "low pass" in nature. In fact, these functional descriptions are very incomplete from an engineering perspective and speaker literature often contains fairly esoteric terms like "first order," "Butterworth," "quasi-fourth order," "Linkwitz-Riley," "Chebychev" and others. These terms describe exactly how the filter attenuates a signal. Not all filters do so evenly. Some begin to act in a very gentle manner, then change characteristics and rapidly reduce output at higher (or lower) frequencies.

Although you needn't concern yourself with most of this, there is one concept you should master – a filter's "order" (or "slope") and its "corner frequency."

Briefly stated, a crossover's "order" is simply the rate at which it attenuates a signal. A "first order" filter does so at 6 dB/octave, a "second order" at 12 dB/octave, a "third order" at the 18 dB/octave, and so forth.

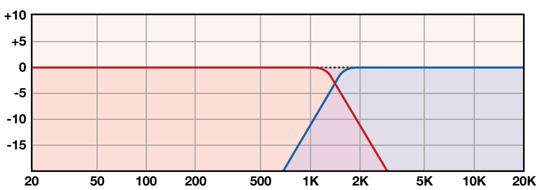
The "corner frequency" (sometimes called the "crossover frequency") is generally defined as the filter's "3 dB down point." Although accurate, this definition is a bit obscure so we'll explain: the "3 dB down point/corner frequency/crossover frequency" is simply the point at which the filter's output is reduced (or attenuated) by 3 dB. This is an important piece of data to an engineer as it often indicates the frequency at which two drivers (a woofer and a tweeter in a two-way design, for example) contribute equally to a certain tone. A woofer would contribute more to lower tones and a tweeter to higher tones but at that particular frequency, the acoustical output of both drivers is equal.



As you can see, this filter does nothing to bass frequencies (it "passes" them, remember?) It begins to affect frequencies around 1.5 kHz – anything higher is attenuated (reduced in level) at the rate of 12 dB/octave. This means that the signal at 3 kHz measures –12 dB compared to the signal at 1.5 kHz. At 6 kHz, it would be reduced by another 12 dB.

From this description, you might surmise that when two drivers contribute equally at that "3 dB down point," the resulting acoustical output would be equal in overall intensity to a lower tone produced predominantly by the woofer and a higher tone produced mostly by the tweeter. You'd be correct. That phenomenon is what we call "acoustical summing." (That accounts for the dotted line in all our crossover frequency graphs).

Here are some practical examples that may make all of this clearer.



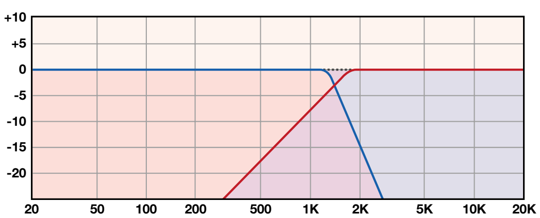
The slope, or rate of attenuation, is greater here. Some engineers prefer higher order filters because they protect drivers better. Other designers claim that they don't sound as good. The jury is still out on this issue.

Let's say you have a second order low-pass filter with a corner frequency of 1.5 kHz. That means the filter would begin to attenuate higher frequencies at the rate of 12 dB/octave.

A third order filter at the same frequency would be similar except for the sharper rate or slope. In this case, the filter's response at 3 kHz would be –18 dB compared to response at 1.5 kHz.

Don't think, however, that all crossovers are *symmetrical* (i.e., have low-pass and high-pass filter sections with the same slope). They often are not.

Some crossover designers prefer to "roll the woofer off quickly," as they put it, with a steep slope (18 dB/octave or greater) filter while bringing on the tweeter much more gently.



Here, the speaker system designer opted to get the woofer out of operation as soon as possible. Note that the tweeter comes into the sonic picture more gently. In this case, the tweeter should have very good power handling capability as it will "see" more low-frequency energy than it would with a sharper slope filter.

Why Is This Important?

It's almost impossible to overestimate a crossover's importance. In addition to basic filtering duties, a crossover provides the only way an engineer can mate a low-efficiency tweeter to a more efficient midrange driver, for example. (The designer may need only to insert a resistor in the crossover's midrange output to compensate.) Crossovers can also be used to equalize a system, to apply more bass, to reduce midrange output, to increase or decrease high-frequency output, etc.

In short, almost all the efforts to "voice" a speaker center around the crossover design. "Voicing," then, is often what distinguishes two similar speakers. It often results in the subtle and sometimes not-so-subtle differences that mark one speaker as truly exceptional and another as "ho-hum."

ENCLOSURES

Don't think that an enclosure is just something to hold the drivers. Enclosures are often as important to the sound you hear as the drivers themselves!

That's because most of the speakers sold today use dynamic drivers and they are more dependent on enclosures to sound their best than are planars.

Enclosures are particularly important for accurate bass reproduction. In fact, the major job of an enclosure is to isolate the front and rear waves coming off the woofer cone and either minimize the effects of the rear wave entirely or control it in some way so that it augments the front wave.

The first "enclosure" was simply a wall. After all, it was the easiest way to totally isolate our competing front and rear waves – one in one room, the other in another!

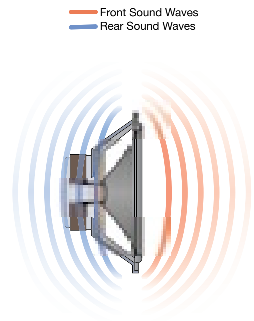
Of course, this approach isn't very practical so engineers quickly came up with less invasive ways to accomplish the same thing.

"Infinite Baffle" and "Acoustic Suspension" Designs

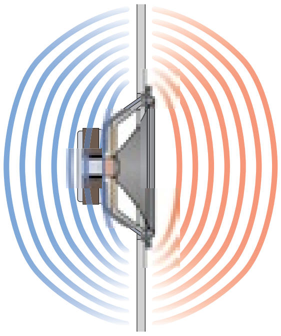
Both of these are "closed box" systems. In other words, there's no way for the air inside the box to get out and no way for the air outside to get in. Although there are some differences between "infinite baffle" and "acoustic suspension" designs due to size, something called "system Q," and the stiffness of the suspension holding the outer edge of the driver, you can think of these enclosure types as closely related. In general, the enclosures of true "infinite baffle" speakers are considerably larger than their "acoustic suspension" offspring.

Almost all enclosures use some sort of material to help control the resonance that develops inside the enclosure. (Sound comes from the rear of a driver, remember?) The object is to *damp* or absorb this back wave, or at least diminish it, so that the internal sound waves don't bounce back to the driver and modulate its output. The point of this material is simply to impede the internal sound waves, to dissipate their acoustical energy by converting it into heat. For that reason, most absorbent material is very loose so that it can trap air molecules easily.

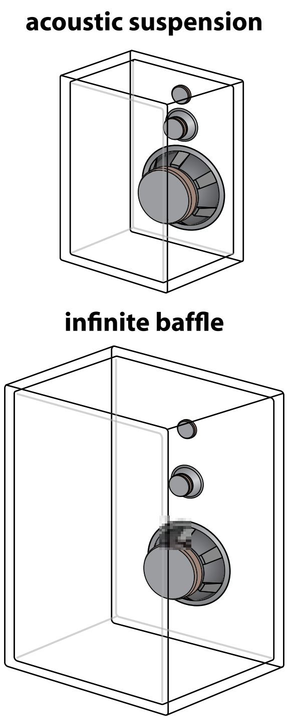
Again, there's no real agreement on which specific material is the most effective. Some high-end manufacturers use combed lamb's wool but most think fiberglass or some other insulating material is just as effective.



If we let the front and rear waves from a woofer mix indiscriminately, they would cancel each other out and we'd hear very little bass.



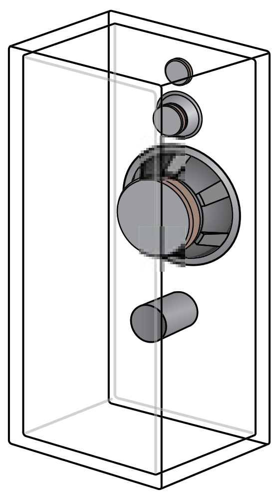
A baffle prevents cancelllation.



The acoustic suspension design quickly won out over its larger infinite baffle parent. Infinite baffle speakers used the large internal air mass as sort of a soft "spring" to complement the comparatively stiff "surround" that holds the cone in place. Acoustic suspension speakers, on the other hand, use their smaller internal air mass as a much stiffer spring. Cone surrounds are, as you might imagine, a bit "sloppier" to compensate.

Acoustic Suspension Pros & Cons

Advantages of the acoustic suspension design include surprisingly accurate bass response from comparatively small cabinets. Thus, acoustic suspension speakers are relatively easy to place in a room. The major drawback is efficiency – or lack thereof. Most acoustic suspension speakers require brawny amplifiers to sound their best.



In a bass reflex enclosure, the driver's rear wave is supposed to escape through the port to augment and extend the driver's low-frequency performance.

Bass Reflex and Passive Radiator Enclosures

These designs are very similar in operation even if they look very different from the outside.

A **bass reflex** enclosure, also known as a "ported" or "vented" speaker, intentionally mixes front and back waves. But it does it in such a way so that the interaction is not sonically destructive.

The trick to properly designing a bass reflex enclosure is carefully matching the driver, the internal volume of the box, *and the dimensions of the port itself* to come up with a combination that works well. (See *Just Who Are These Aussies Anyway?*)

The port is a critical element as its air mass (the volume of air inside the port itself) has inertia and resists the back wave from the driver as the back wave tries to escape the enclosure. When everything's been properly calculated, this resistance helps produce better system bass response than the driver alone is capable of.

Passive radiator speakers operate on the same principle as bass reflex designs but, instead of using the air mass of the port or vent, they substitute a device called a *passive radiator* that looks *exactly* like an active driver. There's a frame, a surround, and what looks just like any woofer cone.

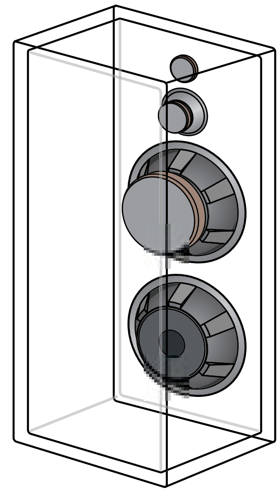
When you look at the rear of this imposter, however, the differences are obvious. For one thing, there's no voice coil, no "former," no spider. And no magnet assembly either.

It's called a "passive radiator" simply because it sits in the enclosure and radiates sound only when the back wave from the active driver hits its own backside. So, the passive radiator design simply substitutes a diaphragm for the air mass in a bass reflex enclosure's port.

Although partisans will argue the merits of each approach until deprived of oxygen, we see little in a functional sense to distinguish them. Both must be carefully designed and tuned for best performance.

Cynically speaking, the one advantage passive radiator designs have is that they're a bit deceptive, particularly to an unsophisticated customer. When the grill is removed, the neophyte might think he's getting another active driver. So read those spec sheets carefully!

Bass Reflex and Passive Radiator Pros & Cons
Bass reflex and passive radiator systems are generally more efficient than acoustic suspension designs. Thus, they'll play louder with the same amp than an acoustic suspension design or they'll require much less amplification to play as loud. Some people claim that ported enclosures are inherently less accurate in the bass end – they either don't have response into the "deep bass" region or their response is "peaky" or "boomy." If properly designed, a vented or passive radiator system is not subject to these flaws.



A passive radiator functions in the same way as its ported cousin.

JUST WHO ARE THESE AUSSIES ANYWAY?

It seems that you just can't have any intelligent discussion about speaker systems without someone mentioning Neville Thiele, Richard Small, or the "Thiele-Small parameters."

What are they? Are they important? (Yes. Very.) Do you need to know anything about them? (No.)

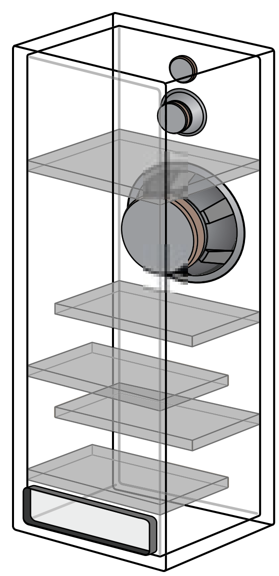
Back in the early '70s, Neville Thiele and Richard Small, both Australians, developed a very valuable set of measurements that allowed a designer to predict a loudspeaker's frequency response before a prototype was even built!

The Thiele-Small parameters, as they're called, involve esoterica like diaphragm area (Sd in engineer-speak), Xmax (not a pagan holiday but a driver's maximum displacement capability), a voice coil's electrical resistance (Re), and about 10 other items.

While groundbreaking in its day, the Thiele-Small parameters are gradually being replaced by an updated series of measurements based on the latest AES (Audio Engineering Society) speaker standards.

The Transmission Line Enclosure

Particularly popular with some British manufacturers, and noted mostly for prodigious and usually quite accurate bass output, the "transmission line" (perhaps more accurately called an "acoustic labyrinth") is a complex enclosure indeed.



More expensive to build than most other enclosures, the transmission line has adherents among those who value extended and accurate bass response.

Although these enclosures allow a bass driver's back wave to reach the outside world, the real benefit is that the long and tortuous path acts more to "load" the driver so that it operates more linearly over its intended frequency range.

Transmission Line Pros & Cons

Most transmission line systems are relatively inefficient and require reasonably powerful amplifiers to sound their best. Another potential drawback is that most transmission line speakers are relatively large. But the rewards, according to their fans, include soul-stirring deep bass that other system types have difficulty matching.

The Missing Enclosure

Although most speaker systems absolutely need an enclosure to perform properly, some don't.

Planar drivers, for example, usually make due very well with just a frame of some sort around the driver itself. The frame adds rigidity but performs no other function – it doesn't "tune" the driver, doesn't control the back wave, etc.

However, you will notice that many planar speakers use an enclosed dynamic driver for bass reproduction. That's because planars just don't cut it down low where drum thwacks and explosions can really irritate the neighbors!

DISPERSION

Now that we've defined driver and enclosure types, it's time to look at **dispersion patterns**, or the ways speaker systems actually project sound into a room.

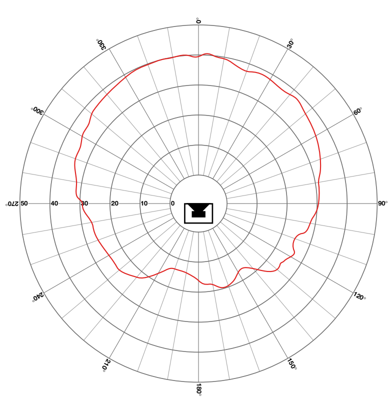
There are only four basic radiation patterns we need to look at:

- monopole (or directional)
- bipole
- dipole
- omni

Monopoles

Most speakers are monopoles. In other words, they project sound in only one direction. (That's why we often call them *directional* speakers.)

Here's a look at how a monopole directs sound into the room.

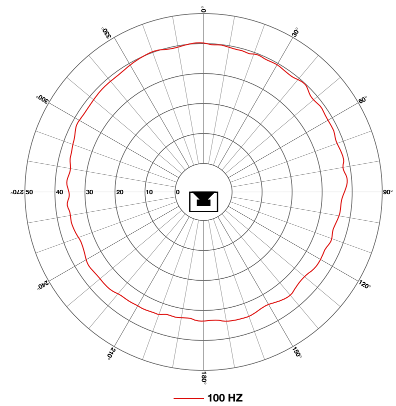


The monopole throws sound in one direction only. So you can "aim" it at just the part of the room you want to.

Unfortunately, though, things aren't quite as simple as our little diagram would have you believe. First, not all frequencies radiate from the speaker system in quite the same way.

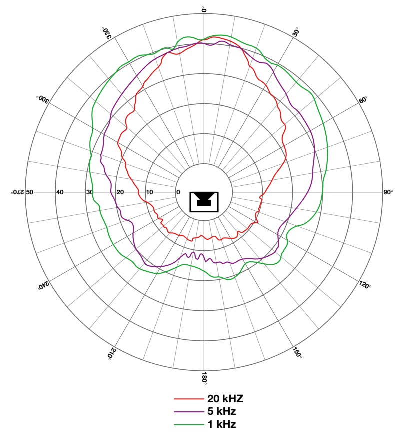
Bass frequencies, for example, usually run willy-nilly to all parts of the room.
But higher frequency radiation patterns are not as expansive. In fact, by the time you're dealing with the two top audible octaves, you need to be sitting so that the speaker is pretty much aimed right at you.

Note: Truly understanding a monopole's dispersion pattern is far more complex than we've indicated here. There's a direct relationship between driver diameter and the wavelength of the frequencies it is reproducing, for example, that determines directionality. Smaller tweeters, for example, radiate sound over a wider area. But a problem arises when a small tweeter may not have a large enough voice coil to handle the signal sent to it.



Bass goes everywhere – or almost everywhere.

There are also questions of driver placement and a phenomenon called "lobing" that also influence radiation patterns. Some speaker designers, for example, use this phenomenon to control dispersion to meet the THX's original and Ultra standards. The so-called "D'Appolito configuration," named after its inventor, Joseph D'Appolito, sandwiches a tweeter between two midrange drivers to aim their combined output more precisely.



High frequencies become more and more directional.

> ***Monopole Pros and Cons***
> ***Monopoles are the most common speaker type. And the vast majority of them use dynamic drivers. Advantages include relatively easy room placement. Despite what some audiophiles might believe, you'll get relatively satisfactory performance by observing a few basic cautions when putting monopoles in a room.***
>
> ***On the down side, monopoles don't easily develop that somewhat difficult-to-define sense of air, space, and delicacy some bi- or dipole lovers cherish. That's because monopoles, by design, generate a more in-your-face perspective. It's a matter of taste.***

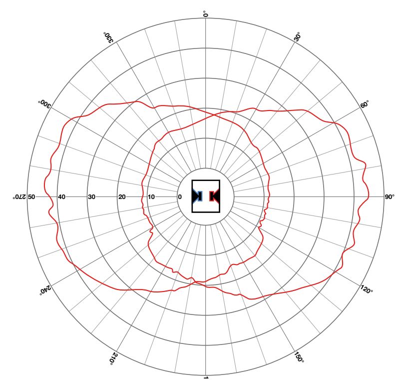
Bipoles

Think of two monopoles back-to-back. That's right, you've got a bipole (or bipolar) speaker.

The bipole's radiation pattern is exactly what you'd expect, too.

Bipoles produce a null to their sides, at least at higher frequencies. As you can see from the polar response chart, they don't generate as much energy to their left and right as they do in the front or back. In other words, they are **not** omnidirectional radiators over the entire audible range.

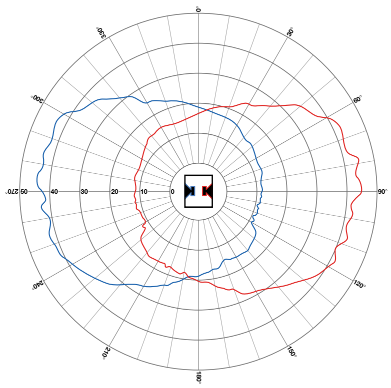
But remember that bass frequencies usually radiate omnidirectionally anyway. That's why bipolar speakers may not have front and rear woofers. Instead, a bipolar's designer may opt for a single bass driver to generate the needed energy.



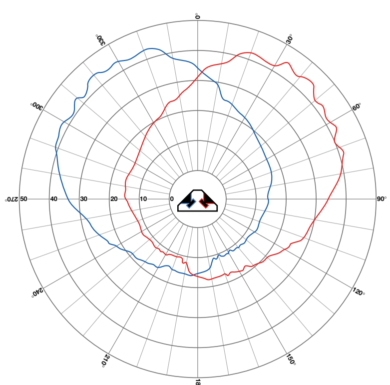
The bipole radiates sound identically in both directions. Front and rear waves are "in synch," or "in-phase:" When the drivers on the front of the enclosure are generating a compression, so are the rear drivers.

Dipoles

Think of an out-of-synch or out-of-phase bipole. There, you've got a dipole!



A dipole's radiation pattern is identical to a bipole's. However, a dipole's front and back waves are out of phase.



Some dipole's used for surround applications have radiation patterns like this.

Dipoles simply radiate front and rear energy that's out of phase. Planar speakers, as we've already mentioned, are dipoles. After all, if a single membrane is moving forward to compress air in front of it, it is, by definition, moving *away* from the air behind it, thereby creating a rarefaction.

Of course, you can have dynamic dipoles, too.

The radiation pattern from most dipoles is very similar to that of a bipole. The difference – and it is an important one – is that the front and rear waves are out of phase when compared to each other.

Remember the null we talked about when discussing bipoles. Well, dipoles, especially dipoles used for surround applications, depend on that null to minimize any sense of directionality that might distract us from the visual and sonic activity in front of us.

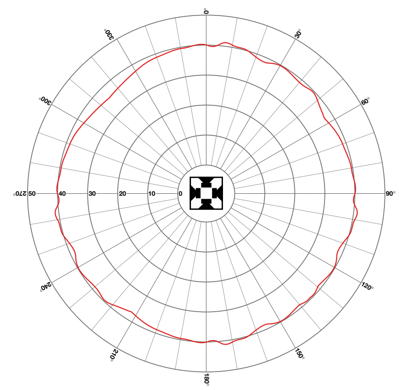
> ***Pros & Cons***
> ***Bi- and dipole zealots point to the laid-back spaciousness these speaker types are famous for. Of course, remember that many of these accolades are earned by planar-driver implementations. It may be difficult to distinguish between driver type and radiation pattern here. By and large, though, there's some truth to these claims, regardless of the type of drivers used to create a bi- or dipolar sound field. .***
>
> ***The dark side, however, is a lack of imaging precision. Bi- and dipolar speakers generally don't generate that "the-guitarist-is-here-and-the-drummer-is-right-THERE!" sense that you can get from a good monopole. The reason? Bi- and dipole speakers bring the room more into the acoustic equation. The back wave, bouncing off your listening room's rear wall before it begins its journey to your ears, may generate a sense of sonic confusion and contribute to a less specific image. And, if your room has some strange acoustical characteristics, bi- and dipoles will probably excite these eccentricities more than monopoles will.***

Omnidirectionals

As the name infers, this is the most "democratic" speaker of all, at least as far as a radiation pattern goes.

"Omnis," as they're affectionately known, generate sound in a full 360^0 spread. Although they can use conventional dynamic drivers (sometimes face-down into an enclosure with the driver's back wave supplying most of the sonic information), most omnidirectional speakers use very exotic drivers indeed. Perhaps the best known is the Walsh driver, named after its inventor, Lincoln Walsh.

Although it first made its appearance in the early '70s, versions of the Walsh driver are still used today. mbl, a high-end German manufacturer, produces omnidirectional speakers based on an unusual design of their own manufacture.



The "omni" does just that – radiates sound everywhere.

Omnidirectional Pros and Cons
If bi- and dipolar speakers can be described as "spacious," the omni wins that race hands down. But there's a price. Omnis will bring your room into the picture as no other speaker type will. Because there's no null, the side walls of your room will reflect a great deal of energy to your ears quickly enough to confuse the lateral image. On the other hand, the front-to-back depth some people experience with omnis is enough to make them think they've gone to sonic Nirvana. It is, after all, a matter of taste.

And the Future?

We don't see much change in basic speaker design coming in the near future. Speakers, for the most part, are the products of fairly mature technology. Although there are several things in labs (multiple lasers with modulating outputs that actually produce sound, for one), we think it's going to be quite a while before we listen to something radically different from the things we've described here.

The one exception is the so-called "flat panel" speaker being developed, licensed, and promoted by nxt and others. These speakers promise to change our basic conceptions of both how speakers work and how they might be used.

For those of us more comfortable with conventional speakers, however, we do see evolving *applications*. Multi-channel movie soundtracks and music reproduction will probably force some of us to reconsider our prejudices and choose more *practical* designs. Computer use will continue to evolve and, with it, speakers.

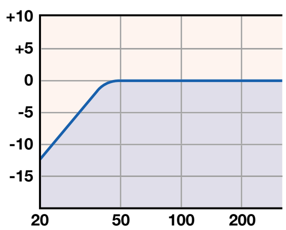
A QUESTION OF RESONANCE

If there's one thing about loudspeaker design that an engineer has to know cold, it's the concept of "resonance." No, we're not about to blast off into Engineering 101 but we are going to discuss some basic ideas.

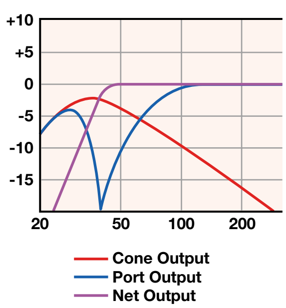
CONCEPT NUMBER 1: RESONANCE

For our purposes, we'll define resonance as the response a physical entity has to sound. Want an example? Tap a heat sink on an amplifier. Listen to the "ring" that persists after we've removed our finger. Tap a speaker enclosure. Listen to the "clunk" or the "thud" or the "tic."

The reason we hear these sounds is because we've excited something called a resonant frequency or, as the dictionary puts it, "the frequency at which the system



*Here, the low-frequency response of the system extends downward very evenly until it reaches system resonance, after which it falls at the rate of 12 dB/**octave**. This simply means that if system resonance was at, say, 40 Hz, the speaker's output at 20 Hz would be 12 dB lower.*



In a vented enclosure, the cone output and the port output augment each other to produce the real system output. This is what you hear. Note that, in a vented system, response below resonance falls twice as fast as it does in a sealed box system or at 24 dB/octave.

would vibrate if deflected once and then allowed to move freely."

Resonance can be incredibly destructive. In fact, the well-known collapse of the Tacoma Narrows suspension bridge over Puget Sound in 1940 was caused by wind-excited vibrations at the natural frequency of the structure itself.

But resonance can also be very good. Speakers, for example, depend on it. So do musical instruments. (The sound box of a guitar, for example, is a resonant system that responds to the sounds produced by the strings.)

CONCEPT NUMBER 2: FREE AIR RESONANCE

All speaker drivers exhibit a "free air resonance." It's the resonant frequency of the driver suspended in free air, well away from any reflecting surface. Free air resonance is determined by the driver's physical parameters: weight, mass, materials, construction, etc.

CONCEPT NUMBER 3: SYSTEM RESONANCE

This is the resonance of the entire loudspeaker system – driver(s) and enclosure. And it's critical to the performance of the speaker.

Here's where these three concepts come together.

Different enclosure types use the concept of resonance differently to achieve their goals. The resonant frequency of an acoustic suspension design, for example, pretty well defines the point at which low-frequency response begins to fall off. Because the box is sealed, the system output and the cone output are identical.

Labyrinth designs, even though they also have a "tunnel to the outside world," as it were, exhibit the same type of low-frequency response as sealed enclosures. In this case, the labyrinth's absorbent material and its complexity combine to emulate the resonant characteristics of a sealed box.

Ported designs are more complicated. Here, we have two sources of low-frequency output – the cone itself and the vent or port that connects the inside of the enclosure to the outside (or, in other words, couples the driver's front and rear waves.)

PREVIEW

If the speaker is the single most important component in a home theater or high fidelity system, the rooms we put them into influence their performance so much we can't really distinguish one from the other in any practical way.

In fact, if you wanted to change the way a system sounds, just move the speakers about 6"! That's right. Moving speakers just half a foot will change how they interact with a room so much that you'll hear more of a difference than you will with any amplifier swap or change in cables. It's that simple.

So we'd better know our rooms and why they do what they do. This Section will cover a bit of basic acoustics with an emphasis on how rooms influence our perceptions. We'll also investigate how speakers and rooms interact and suggest some placement choices for speakers that should help you realize the best audio performance a system is capable of. And, of course, we'll emphasize some "real world" tricks that will make that often-daunting task a lot easier.

SOME BASIC ACOUSTICS

There's a major transition when theater comes home. It's space — or the lack of it. Simply stated, rooms in the homes most of us live in are smaller than theaters and we need to draw some lessons from that obvious fact. Before we do, let's take a look at some "theater facts" for sound systems:

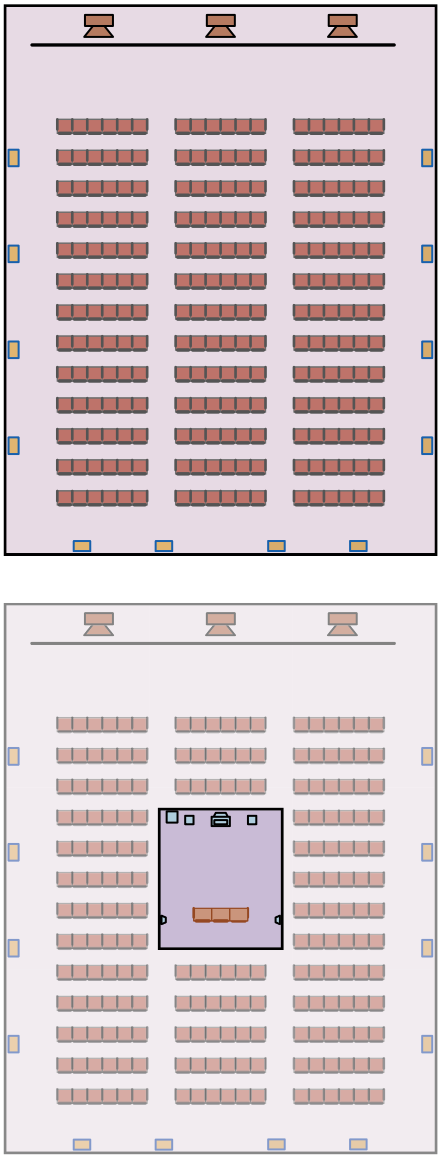
Mid- and high-frequency coverage must be very wide or a lot of the audience simply won't hear everything the film's director and sound engineers want them to.

This "even" sound spread is very difficult to achieve as the audience is widely spaced about the theater with most individuals sitting some distance from the screen.

Effective low-frequency effects (rumbling trains, explosions, volcanic eruptions, etc.) are very demanding. The sound system must move an extraordinary amount of air to do this well.

In addition to these broad problems, a theater's sheer size provides additional acoustic challenges, the largest of which is the negative impact on soundtrack clarity caused by early reflections and reverberation.

Most acoustical considerations are fairly complex and outside the scope of this book. However, you'll need a bit of background on the impact of room reflections as they come into play every time you set up a home theater or music system in a customer's home.



The size difference between a theater and a room in our home means that the acoustical design considerations are not always the same even if the final goal of getting good sound to everyone in the room is.

"Dead" Rooms vs. "Live" Rooms - Which Is Better?

Before we take a closer look at room reflections, however, we need to dispel one myth that seems to perpetuate itself whenever anyone begins to take even a casual look at the science of acoustics – the myth that states that the best place for the reproduction of sound is an anechoic chamber (or, more simply stated, a totally "dead" room with no reflections, no echoes, no reverberation, no standing waves, etc.)

The argument goes something like this: If the sound source contains everything the artist/producer/engineer wants us to hear, why muck it up by playing it in a room that imposes its own "sound" on the signal? Wouldn't it be better to play it in an environment that just lets the loudspeakers "speak"? In other words, isn't a "live" room with its complex patterns of reflections, echoes, reverberations, standing waves, etc., detrimental to our appreciation of a previously recorded event?

In a word, the answer to that is a resounding "**NO!**" Although intriguing on the surface, this idea is seriously flawed. An anechoic chamber (a room with no reflections or reverberations) is a disconcerting place in which to do anything, let alone something as potentially enjoyable as listening to music or watching a movie. The reason is that the very reflections that "change" sound somewhat also give us a comforting sense of sonic orientation. Without them, we feel uneasy.

Try this: if you have a cramped closet filled with clothes, walk in, close the door, and just sit for a few moments. If the closet actually has enough clothes in it to absorb sound, you'll soon begin to feel very uncomfortable. In fact, you may even begin speaking to yourself — and then you'll feel even more uncomfortable because you won't hear the familiar reflections you're used to. Imagine that effect increased by several orders of magnitude and you'll begin to appreciate what an anechoic chamber is like — maybe a nice place to visit but you certainly wouldn't want to live there.

The opposite of an anechoic chamber is the so-called "reverberant chamber" in which nothing ever dies — sonic reflections seem to linger forever. It is the epitome of a "live" room and is as equally disconcerting as its "dead" anechoic relative.

Thus, the answer to the "dead" vs. "live" controversy lies, as you might expect, somewhere in the middle. As it turns out, the best rooms for viewing and listening are those with enough reflections to allow us to feel comfortable but not so many or so uncontrolled that they seriously change the tonality or the spatial characteristics of the source we are trying to reproduce.

Where Do Reflections Come From?

Even the most directional loudspeaker spreads sound over a fairly wide area. This spread causes room reflections as speaker output hits boundaries like walls, a ceiling, etc., and bounces back to our ears.

Room reflection problems generally impact the intelligibility and "naturalness" of a signal. In the theater, long delays between the arrival of direct and reflected signals aren't as much of a problem as they are in a smaller room. (The reason for this is a bit complicated but it centers on our ability to subconsciously disregard identical signals if they're spaced far enough apart in time.) In a small room, reflections usually produce a type of sonic confusion that robs the signal of detail.

> *Reverberation, or the quality of an enclosed space that supports a sound after the source stops, is a major related factor. It, too, "smears" transients and affects intelligibility but in a different manner. Fortunately, reverberation is a phenomenon associated with larger spaces than we usually find in the home.*

This is why, for example, THX loudspeaker specifications call for comparatively narrow vertical dispersion patterns that minimize floor and ceiling reflections. On the other hand, these same specifications call for a very *wide* horizontal dispersion pattern for optimal coverage. This may aggravate problems caused by wall reflections but, fortunately, these wall reflections are easier to deal with than are reflections from floors or ceilings.

Another variable you need to consider is the nature of the signal you are playing. Our perception of stately, classical music with a lot of low-frequency information (cellos, bassoons and the like) is affected far differently by time lags between direct and indirect sounds than is our perception of quick, percussive music — or speech, for that matter. As movie soundtracks usually combine every variety of audible sound, they present a particularly difficult signal to deal with.

Although smaller rooms (a home theater room, for example) don't suffer these acoustic anomalies on the same scale as do large rooms (the proverbial cathedral, et. al.), you still need to guard against the destructive influences anyway. If you don't, your customer will experience a loss of intelligibility that may not be glaringly obvious at first but will, over the long haul, decrease their enjoyment.

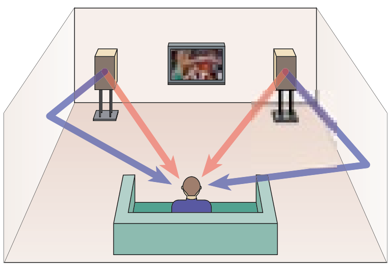
You can prevent many of these problems in two ways:

1) **Absorption**: A sound wave can't reflect back into a room if it has been absorbed (in reality, converted into heat) by something. That "something" may be a drape or a heavy fabric tapestry hanging on a wall or an acoustical panel, etc., that impedes the motion of air molecules as they convey the sound wave. Not all materials are created equal in their acoustical effectiveness, however.

For example, heavy fabric drapes hung several inches out from the wall are far more effective in absorbing sound waves than sheer curtains very close to the wall.

Each material has a different "absorption coefficient." Remember that this ability is measured by:

- the **amount of attenuation**
- the **range of frequencies** most affected. Because each material is a bit different, you'll need to consult an acoustical textbook for exact figures.



The indirect sound waves are delayed by fractions of a second simply because they travel longer distances. The amount of delay has an impact on the apparent clarity and directionality of the original signal.

As unspecific as this might sound, the following guidelines will help:

- Thick, loosely woven fabric will absorb a wider range of frequencies and do so more effectively than will thin, tightly woven fabric.
- The effectiveness of any material increases dramatically when you hang or place it well away from the wall. This goes for acoustical panels as well as drapes and curtains, etc. That's simply because the increased fabric-wall spacing helps reduce the intensity of the reflected wave more effectively.

2) **Diffusion**: Diffusion, or the breakup of a primary soundwave into smaller, less intense waves, can also benefit a home theater room. Diffusion is not all that difficult to achieve, although proper diffusion techniques for a particular room may be more problematic. General rules are as follows:

- Irregular surfaces break up sound waves more effectively than large flat areas.
- The type of diffusing surface is important.
    - Concave surfaces tend to focus sound waves rather than diffuse them. Avoid concave surfaces whenever possible.
    - Convex surfaces are very diffusive, comparatively easy to implement and are generally recommended.
    - Complex diffusors (actually 1/4 wavelength diffraction gratings) made by companies like RPG Diffusor Systems, Inc. of Largo, MD, are extraordinarily effective but comparatively expensive. They should be specified and installed in cooperation with an acoustical consultant.

A more practical application of the same principal might occur if your customer's home theater system was going into a room with a lot of books. Try pushing all the books to the back of the shelves rather than lining them up flush with the outer edge. Books have different depths so the resulting irregularities may just provide an inexpensive yet fairly effective means to diffuse sound waves.

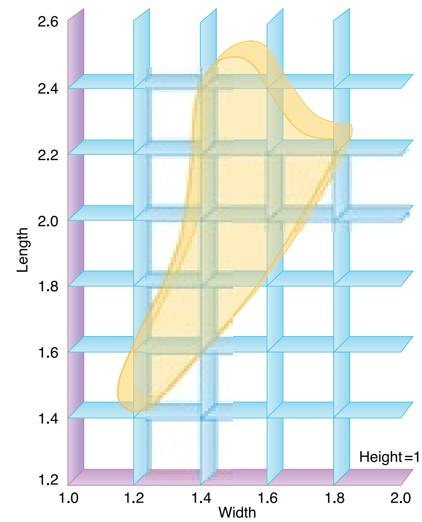
Room Dimensions and "Standing Waves"

The actual dimensions of a room have a profound influence on the sound quality of that room, particularly in the low-frequency area. This has to do with so-called "room modes" (also called "standing waves") that cause the room to respond in very uneven ways to equal intensity signals of different frequencies.

A full explanation of this phenomenon would take several pages and is based on the fact that rooms are actually resonant systems — much like loudspeakers. The explanation would cover the three common standing wave patterns or room modes called "Axial," "Tangential," and "Oblique."

When you consider that a loudspeaker in a room is a resonant system within a resonant system, you'll begin to appreciate the complexities. A full explanation of all the forces at work here is guaranteed to turn your eyes plaid so we'll give you some "quick and dirty" guidelines to go by:

- Avoid cubical rooms like the plague!
- Square rooms aren't much better.
  - These are acoustical horror stories waiting to be written. Don't saddle yourself with systems in these rooms if at all possible.
- Rectangular rooms are much better than square rooms but, even here, **the ratios of length to width to height are very important**. The available literature does not totally agree on the "ideal" ratios but the following chart gives you a feeling for those generally accepted as "safe."



The room dimension ratios covered by the area inside the dotted line are generally accepted as optimal for the even distribution of low-frequency room mode patterns. This area is called the "Bolt area" (after the studies by R. H. Bolt published in the Journal of the Acoustical Society of America.)

Remember that rectangular rooms offer several advantages:

- There are far more of them!
- They usually provide more and better sightlines to the screen
- The acoustical properties are more easily understood
- Satisfactory speaker locations are usually easier to find.

Some closing thoughts:

- No room is perfect, even those specifically designed for excellent acoustics.
- Rectangular rooms usually work best for home theater or music-only systems providing some elementary care has been taken to control unwanted reflections.
- The actual dimensions of a room are not as important in determining sound quality as are the ratios of those dimensions: Rooms with dimensions that are exact multiples of each other usually do not sound as good as rooms where the dimensions are not directly related.

SPEAKER CHOICES FOR HOME THEATER

Now that you've gotten a taste of some of the acoustic properties you must deal with, let's look at loudspeakers and how to most effectively place them for maximum enjoyment.

What You Need

Home theater is obviously a multi-channel experience. But it is, nonetheless, ONE experience. That has some implications for the speakers you select for a particular system. Let's look at what each speaker's role is and how to determine what to present to customers.

Left and Right Speakers: These speakers handle a great deal of the soundtrack's information and are critically important for overall realism.

- All the considerations applied to high fidelity speakers apply here: **Even frequency response, low distortion**, etc., are essential.
- Because of the dynamic demands of most movie soundtracks, home theater speakers should be **efficient and capable of substantial output** when necessary.
- In addition, you should recommend only those speakers (especially center-channel speakers) with **magnetic shielding**. If the system you're planning involves a direct view TV, this is mandatory! Rear projector sets are also vulnerable to unshielded speakers, front projectors much less so. Shielded speakers prevent stray magnetic fields from interfering with the picture or possibly permanently damaging the TV.

    This magnetic interference comes from the flux lines generated by the magnets on each driver inside a speaker enclosure. Shielding is actually easy: a designer can simply glue a second magnet to the rear of each driver's active magnet. The second magnet focuses the flux radiation and prevents spurious radiation. This is the approach taken by almost all manufacturers as it is both effective and comparatively inexpensive. Another approach involves covering the entire magnet with a magnetically impervious substance known as mu-metal. This is very effective but exceptionally expensive and therefore rarely used.

The Center Channel Speaker: Perhaps the most important speaker in a home theater system, the center channel handles everything from earth-wracking explosions to whispered dialog. For that reason, the center channel should at least equal the Left and Right speakers in all performance characteristics.

- It should be capable of **extraordinarily dynamic performance**. It is responsible for the intelligibility of all dialog information, even when that dialog is accompanied by explosions, earthquakes and other elements that might otherwise obscure accurate reproduction of the spoken (or shouted!) word.
- The center channel speaker **should have the same tonal quality** as the Left and Right speakers. This will prevent sudden sonic discontinuities as a scene

pans across the screen and the sound follows from one side to the other. After all, you don't want that Ferrari momentarily turning into a Lamborghini and then back into a Ferrari again as a scene pans from left to right, do you?

- As we've already pointed out, the center channel speaker **MUST be magnetically shielded** as it is designed to be placed very close to a screen.
- **Practicality is important**: You wouldn't suggest a $1,000 center channel speaker for an inexpensive system. On the other hand, a truly good system needs the best center channel speaker available.

Occasionally, you'll have to match a new center channel speaker with existing Left and Right channel speakers. This isn't easy but there are some guidelines you can follow.

- Determine what make and model the Left and Right speakers are. **If they're recent models *and shielded*, use another of the same model.** This will take care of all the questions about matching tonality, won't it? However, you'll be left to seek another solution if the speakers are old, out-of-production models.
- **Investigate a dedicated center channel speaker made by the same company that made the main speakers**. This is much more a "real world" scenario and is comparatively easy to do. The advantage is that the common design heritage probably means the new center channel speaker will have a tonal balance similar to the existing speakers. If so, "cross-screen" pans will be free of any disconcerting change in sound quality.
- **Use a center channel speaker made with similar drivers to those used in the main speakers**. Although attractive in theory, this has limited "real world" usefulness as speaker "voicing" (the designer's careful blend of driver characteristics, crossover details, and enclosure design to produce a certain sound) will undoubtedly be different. However, you may still find an attractive match.
- After comparative auditioning, **decide on a center channel speaker that shares no common heritage at all with the main speakers.** This is probably the most common scenario. However, the sonic differences may well point out that a whole new LCR array is needed. If this is the case, the existing speakers can be used in another room.

Surround Speakers: Choosing surround loudspeakers may seem a far easier task than picking LCR speakers. Be careful here. Yes, it's true that older Pro Logic (but not the new "II" variant!) decoding produced a somewhat anemic surround signal. However, today's digital soundtracks – not to mention surround music! – can tax surround speakers unmercifully. For that reason, surround speakers need:

- **Very even response** throughout their useful frequency range. Cheap speakers have substantial variation through their passband and won't be satisfactory.
- **The dipole question**: to THX or not to THX?: An early choice here will help you focus on appropriate models.

Once you've worked through these considerations, here are two more important notes:

- **Surround speakers need the same tonal characteristics as LCR speakers for smooth front-rear pans.** Surround speakers with wildly different sound qualities will prove disconcerting as their sonic discontinuities simply won't allow an enveloping soundfield to develop in the first place.
- **Surround speakers do not need magnetic shielding** as they're always located far away from the TV screen.

Choices for Surround speakers appear almost endless. In theory, any small, easily mounted speaker can serve. In reality, choices are restricted by how well the surround speakers mesh with the LCR speakers. The easiest way to insure a satisfactory front/rear blend is to choose surround speakers from the same manufacturer who supplies the LCR speakers.

Subwoofers: Subwoofers are loudspeakers specifically designed to accurately reproduce the two or three lowest musical octaves, or from 20 Hz up to 80 or 160 Hz. Most home theater systems limit low-frequency output to 80 or 100 Hz in order to avoid the "bass localization" effects that take place when a subwoofer's response extends higher than that cutoff point. In general, subwoofer performance is measured by conventional standards:

- Quality subwoofers produce **output to below 30 Hz** without significant drop-off and show **very even frequency response** to their upper limits.
- **Distortion components must be kept low** (a surprisingly difficult task at bass frequencies) in order to maintain the audible clarity of even *very* special effects.
- **High efficiency and dynamic capabilities are vitally important** as home theater subwoofers must deal with the extensive demands today's soundtracks place on low- frequency reproduction.

From a functional point of view, there are two types of subwoofers: passive (non-powered) and active (powered).

- **Passive subwoofers** (rarely seen today) consist solely of a driver (or drivers) in an enclosure. They're designed to be used with an external amplifier. Some may include an internal passive crossover to limit high-frequency signals from getting to the voice coil(s) of the bass transducer(s), some may depend on an external high level (or electronic) crossover placed upstream of the dedicated power amplifier.
- **Active subwoofers** (by far the most common today) add an integral crossover and power amplifier to the driver/enclosure module. Most manufacturers incorporate the electronics inside the enclosure for convenience while a few others provide a separate chassis for the electronics. The exact configuration is a matter of choice as neither one provides a theoretically significant performance advantage.

How many subwoofers does a home theater system need?

That depends. In theory, you can get away without any. Remember that most processors' initial set-up screens ask "Do you have a subwoofer?" If you answer "No," the processor's Bass Management circuit directs bass information to the wide-range speakers and you'll most likely hear substantial low-frequency output.

However, most home theater systems include at least one subwoofer. You might even suggest a second sub connected to the same LFE output if your room is exceptionally large. The nature of bass propagation (or how rooms support bass reproduction) is such that two subwoofers, even if they're reproducing exactly the same signal, can generate a significantly smoother sound than one. In addition, some experts are now recommending a separate subwoofer for the rear channels only. This may help augment the smaller surround speakers' reduced bass output capability.

> *The final choice is obviously up to you. However, we strongly recommend a minimum of one subwoofer. It will provide a more exciting and realistic foundation to the sound of any source you play through that system. Consider two subs in exceptionally large or difficult rooms. An additional "surround-only" sub may be an attractive option for some system configurations.*

Remember that placement requirements for good bass response and mid/ high frequency reproduction are usually very different. That's one of the main advantages of a separate subwoofer. You'll soon see more about this when we discuss placement options shortly.

WHERE DO THE SPEAKERS GO?

Here are the broad guidelines for home theater loudspeaker placement. These recommendations are NOT "cast in stone" as almost every system and every room has some eccentricity that requires a bit of modification. However, if you stray too far from these guidelines, you will probably create more problems than you'll solve.

1) Left and Right Speaker Placement:

These speakers should flank the TV screen and subtend an angle of 45° when viewed from the main viewing/ listening position. Most loudspeakers sound best when placed away from the front and side walls of the home theater room.

Make every effort to place both speakers on the same plane. This means they should be at the same height as each other, preferably with their tweeters at ear level when you're seated in the prime location. They should also be placed at the same distance from that prime viewing/listening location.

Place the Left and Right speakers so they form a 45° angle when viewed from the viewing/listening position. This assures optimum integration of the visual image and accompanying sound and is independent of screen size.

Note that there are some speakers specifically designed for in-wall or on-wall placement. If you're using these speakers, follow the manufacturer's recommendations.

What About "Toe-In"? Toe-in is a simple technique that sometimes makes a significant difference in the way a speaker pair projects a convincing image. Toe-in simply means aiming the speakers inwards a few degrees so that the tweeters of the stereo pair are aimed more precisely at the prime viewing/listening position. Toe-in's effects are variable depending on both the speaker and the room setup. The only advice we can give is to experiment. You may experience significant improvement or no difference at all. Either way, the cost of this "tweak" is certainly minimal – just a moment or two and some enjoyable listening!

Note*: If the system is intended more for multi-channel music listening than for movie watching, place the main speakers so there's a 60° spread between them rather than the narrower 45° degree spread mentioned above. The 45° angle is ideal for movies, the 60° spread better at conveying a wider, more natural musical soundstage.*

2) Center Channel Speaker Placement:

The center channel speaker must be as close to the TV screen as possible. Of course, different display types may impose some different placement options. Here are some suggestions we've found helpful.

a) For Direct View sets

- Place the Center Channel speaker either immediately above or immediately below the screen. Ideally, the tweeter of the center channel speaker should be at the same height as the tweeters of the Left and Right speakers. Choose the location (above or below) that places the tweeter closest to an imaginary line drawn between the tweeters of Left and Right speakers. Make sure you "aim" the Center Channel speaker at the seated ear level of the primary viewing/listening position.

- The distance between the Center Channel speaker and the main viewing/listening position should be exactly the same as the distance from the Left and Right speakers. In practical terms, this means that the Left, Center, and Right front speakers should form an arc in the following manner:

b) For Rear Projector sets

- The spatial (relative height and distance) placement requirements are the same as with a direct view TV. However, a rear projector TV often has no place for a Center Channel speaker below the screen so top placement may be mandatory.

- If the chosen Center Channel speaker has a rear vent or port, do not place it in a tight compartment in the projector cabinet. The cabinet may "load" the speaker, change its tuning, and alter the sound quality.
- Remember to aim the Center Channel speaker at the prime viewing/listening position.
- Most rear projector sets are so large that equidistant placement of Left, Center and Right speakers is difficult. For that reason, you may have to keep the Center Channel speaker as close to the front edge of the cabinet as possible.

> **Note*: You may be tempted to push the Center Channel to the rear of the cabinet to make the arrival times from each speaker approximately equal.* Resist that temptation. *The large flat cabinet top immediately in front of the speaker will do more harm to the resulting sound than the slight difference in arrival times.***

- If the rear projector set is very large or has a cabinet style that makes it impossible to position the center channel speaker at the proper height, consider using *two* center channel speakers, one mounted immediately above the screen and the other immediately below.

  This will create a "phantom center" speaker that will appear to originate from a height between the two real speakers. If done properly, this will insure that center channel information will "come from the middle of the screen."

> **Note*: If you have to use this technique, make absolutely sure that the two center channel speakers are equidistant from the main viewing/listening position. This may mean you'll have to hang the top center channel speaker from the ceiling.***

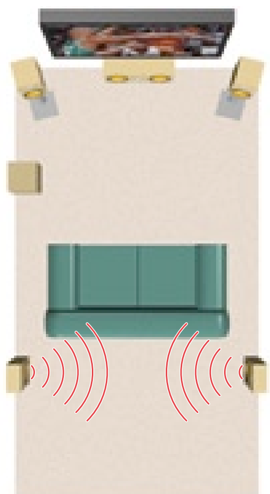
c) For Front Projector (two piece) sets:

- Use an acoustically transparent screen set whenever possible. This allows very flexible center channel speaker positioning, at least as far as relative height goes. It also provides the best synchronization between image and sound as all dialog (usually the most important portion of center channel information) appears to be coming directly from the screen. Make sure you **don't** place the center channel speaker in close proximity to the screen's rear surface. Exact distances will vary with screen type: Consult the screen manufacturer for exact recommendations.
- If an acoustically transparent screen can't be used, follow the guidelines for direct view or rear projection sets.

d) For Flat Panel displays

- The size of the display is the biggest factor in placing a center channel speaker.

In general, we recommend following the steps outlined for Rear Projector sets above. However, some installations may call for even more imaginative solutions.



If you'll be listening primarily to movie soundtracks, place your surround speakers on the side walls just behind your main listening/viewing position.

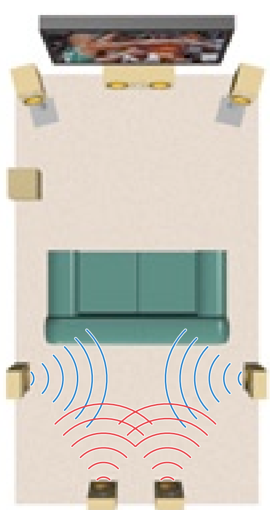
3) Surround Speaker Placement:

This is one of the more challenging areas of home theater system design. Placement requirements will change depending on the type of speaker chosen and the room itself. Although surround speakers could be smaller than the main speakers in the past, this is not necessarily true today. In fact, contemporary soundtracks occasionally place extraordinary demands on a surround speaker's bass output capability. And if the system will be used for multi-channel music reproduction, there's no telling what an artist or producer will place in the surround tracks.

With movies, however, remember that surround's goal is to place you in an enveloping soundfield without adding another easily identifiable and distracting "source" for that information. This makes proper placement all the more important. The following guidelines will give you a place to begin.

a) Monopole/direct radiating surround speakers in a "mostly movies" 5.1 system:

- Place them high on the side walls (i.e., well above ear level) and slightly behind the main viewing/listening position so that their radiation pattern avoids excessive localization that often proves distracting while watching a movie.
- Avoid aiming them directly into the prime viewing/listening area. In fact, in some cases it's better to "bounce" surround information off a rear wall before it reaches the viewing/listening area.
- You may need to add some acoustical treatment (primarily diffusion) to the area around the surround speakers. This is particularly true if you're bouncing sound off adjacent walls.
- Remember that experimentation is the key here. You may need to move or re-aim the surround speakers for the most enveloping rear soundfield.



A "6.1/7.1" system adds speakers at the rear of the home theater room to reproduce "center back" information contained in whatever source you are playing.

b) Monopole/direct radiating surround speakers in a "mostly movies" 6.1/7.1 system:

> **Note*: Most "6.1/7.1" systems add speakers to the rear of the room (and slightly more complex electronics, of course) to provide a "center back" channel that places some surround information directly behind the audience* in addition *to that already coming from the regular surround channels of a normal "5.1" system. This can improve the sense of envelopment we normally experience from just two surround channels. In addition, the center back channel gives movie directors and sound engineers a better way to add very specific spatial information to the soundtrack when they need to do so.***

- Mount the additional "center back" speakers high on the rear wall at approximately 1/3 intervals. Thus, the left "center back" speaker should be placed about 1/3 of the way across the wall from the left corner while the right "center back" speaker will be best placed at the 2/3 mark.

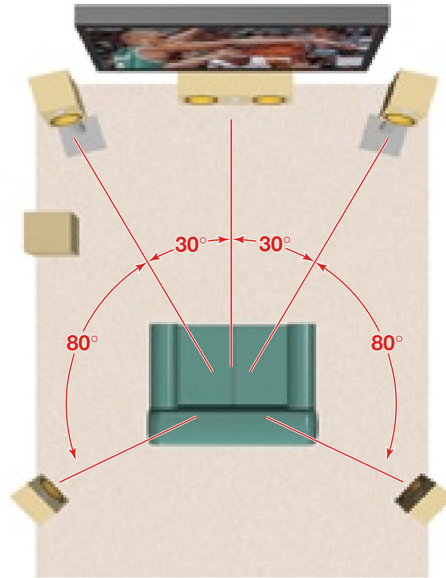
c) Monopole or direct radiating surround speakers in a "mainly music" 5.1 system:

Most music producers and recording/mastering engineers prefer monopole speakers for all "wide range" (i.e., L, C, R, LS, and RS) channels. In fact, the preferred speaker configuration is one where the speakers used to monitor these channels are identical. This means that the differences they hear are being produced by the source (or their manipulations of it) rather than by any speaker differences. If multi-channel music is your goal, we strongly recommend that you follow their lead and select and place your speakers accordingly.

As we've already indicated, a 60° spread between the Left and Right front speakers is best for music. The center channel speaker would be, naturally enough, directly in front of the main viewing/listening area with the Left speaker 30° to its left and the Right speaker 30° to its right.

- International standards for multi-channel music monitoring call for surround speakers placed ± 120 ° from the center of the front array per the diagram immediately below.

- In addition, the surround speakers for "mostly music" systems need not be placed as high on the walls as they might be for a "mainly movies" system.



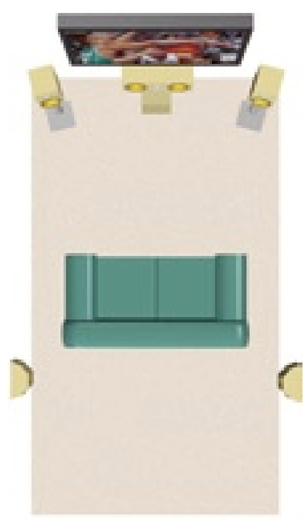
This diagram shows the international standard for speaker placement in a multi-channel music system. Note the comparatively wide surround speaker placement. This makes sure that ambience information is widely dispersed. It's also an acknowledgement that producers and engineers rarely place direct instrumental or vocal sounds behind us unless the music is specifically scored for this type of antiphonal reproduction.

d) Monopole or direct radiating surround speakers in a "mainly music" 6.1/7.1 system:

To date, we haven't seen a music source in this format. That doesn't mean we won't but we suspect (hope?) that that event will be far, far, *far* in the future.

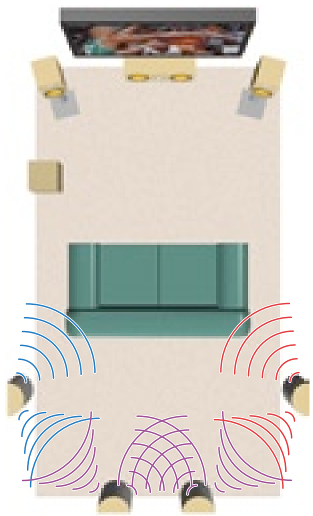
e) Dipole surround speakers in a "mostly movies" 5.1 system:

- As with direct radiators, mount dipole surround speakers high on the side walls just to both sides of the viewing/listening area.
- Make sure that the dipole's "null" side (the outer side of the enclosure without any drivers on it) is aimed directly at the main viewing/listening area. This insures that the speakers will distribute sound so that very little of it reaches the prime viewing/listening area directly. In general, dipole surround speakers do not need any acoustical treatment around them if they are properly placed.

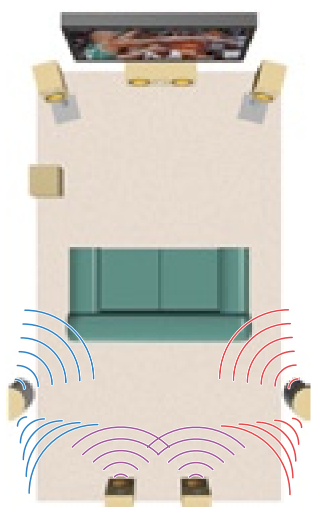

Dipole speakers create a very spacious but non-specific feeling of ambience.

f) Dipole surround speakers in a "mostly movies" 6.1/7.1 system:

This is the "Surround EX/ES" configuration with "center back" speakers added



An "all dipole" 6.1/7.1 surround array.



A mixed "dipole/monopole" 6.1/7.1 surround array.

to the usual dipole side-wall surrounds. If you elect to use dipoles for all surround speakers, note that you should position the dipoles per b) above. However, note that dipoles are marked for polarity as one set of drivers will produce an initial compression pulse while the other driver set will produce a rarefaction with the same input.

g) Hybrid surrounds (two dipoles, two monopoles) for "movie or music" 6.1/ 7.1 system

This may be a good answer for many listeners who will use their home theater system for both movies and music. The setup is very similar to f) above except for the substitution of monopoles in place of the rear wall dipoles.

4) Subwoofer Placement

We've left the best to last as **subwoofer placement is probably the most critical and the most misunderstood element in setting up a home theater.**

This is due to two elements:

1) most people want lots of bass for their special effects but then complain that their music sounds muddy and
2) good bass reproduction is not as placement-independent as some folks would believe.

The first is a matter of subwoofer/wide range speaker balancing while the second really gets into the matter of where to put the subwoofer. These elements are interrelated and trying to deal with one without considering the other will not give a satisfactory solution to the problem.

Rather than detail all the possible considerations, remember this general rule: A room's standing wave or room mode pattern determines our perception of sound. Many room mode peaks occur at a room's boundary junctions (the point where a floor meets a wall, for example) and all room modes peak in the corners. Those facts give rise to the following generalizations:

- You'll get the *least* amount of bass with a subwoofer placed towards the center of a room.
- You'll get *more* bass with a subwoofer placed closer to a boundary junction.
- You'll get the *most* bass with a subwoofer placed in a corner.

Notice we've talked only about the *quantity* of bass output so far. We haven't made any comments about the *quality* (or evenness) of the bass response. And we haven't said that "corner placement gives a 6 dB increase in bass response" or something to that effect. The reason we haven't used specific numbers is that each room's standing wave pattern is so unique that such a statement would be, at best, misleading.

The real trick to proper placement is to find that point in a room where the output of the subwoofer excites the resonant pattern of the room as evenly as possible (i.e., generates the "flattest" frequency response **as measured at the main viewing/listening position**.

The normal method for finding this elusive spot (it IS different for every room – and for every viewing/listening position in that room!) is very time consuming. It involves placing the subwoofer in a number of different locations, using a random test signal to excite the air mass of the room and then measuring amplitude response variations at the viewing/listening position. You place, you measure (looking for the flattest response of course), you place, you measure (looking for the flattest …), you place again, you measure again, …

Eventually, you tire of the whole thing and settle for the best position you've discovered so far. HOWEVER, if you remember that rooms are resonant systems and that resonant systems are not usually "one way" devices, you might think of reversing this process.

Instead of moving the woofer constantly and measuring the results at one place, why not put the woofer in the viewing/listening position and measure the room at various spots? Of course, this procedure still requires a real time analyzer for maximum effectiveness but otherwise it is an elegantly simple solution to a complex problem.

To locate the best position for a subwoofer, then, follow these steps:

a) Put the subwoofer on the chair or sofa you've decided will be the prime position in the room. Make sure that the driver (and port if the speaker is a reflex design) or slot is as close to ear-position as is possible. (Of course, this is only temporary! You'll take the subwoofer off the furniture as soon as you're done! Right? ***Right***?)
b) With a bandwidth-limited pink noise source (the test tone generator included in almost every surround processor for example) playing through the subwoofer at 80-85 dB SPL, walk around the room with a real time analyzer and look for the position that registers the flattest response on the RTA. (Remember that you want the analyzer's microphone close to where the subwoofer's driver/slot/port will be when it is placed on the floor.)
c) Place the subwoofer in that spot.
d) Verify the response with the RTA at the viewing/listening position.
e) Tweak as necessary.

It's easy when you know how, isn't it?

A FEW COMMENTS ON THX-CERTIFIED SPEAKER SYSTEMS

THX's original speaker specifications (and the first iteration of the Ultra specifications) incorporate many of the points contained in the paragraphs above but either extend the performance requirements or throw a different slant on them to bring them closer to how THX thinks a home theater system should perform.

Here's a list of the more important THX specifications that apply to loudspeakers:

1) Left, Center, and Right Speakers:

- **Wide horizontal dispersion** (60°) to insure that the front soundfield is evenly spread over the entire viewing/listening area. This significantly extends the "sweet spot" (the area of full dimensionality) to more people in the home theater room.
- **Controlled vertical directivity** (Directivity Index of 8 dB) to minimize ceiling and floor reflections that might otherwise interfere with soundtrack intelligibility.
- **Sensitivity** (>89 dB SPL) to insure adequate efficiency to accurately reproduce film soundtracks with commonly available amplifiers.
- **Continuous power handling** (100 dB SPL in 3000 ft^3 room for 3 hours): to guarantee reliability and sufficient dynamic headroom for uncompressed reproduction of dynamic peaks.
- **Magnetic shielding is mandatory**.

2) Surround Speakers:

- **Dipolar radiation pattern**: to insure adequately diffused projection of surround information into the viewing/listening area.
- **Flat power response** (125 Hz - 10 kHz ± 3 dB) to guarantee accurate tonal balance between front and surround soundfields. (Note that "Power Response" is NOT the same as "Frequency Response" as it is the average of many off-axis measurements and is therefore a far more demanding "real world" measurement.)
- **Sensitivity** (>83 dB SPL): to insure adequate efficiency to accurately reproduce the ambient and effects information contained in film soundtracks.
- **Continuous power handling** (97 dB SPL in 3000 ft^3 room for 3 hours): to guarantee reliability and sufficient dynamic headroom.

3) Subwoofers:

- **Frequency response** (35 - 200 Hz, +0, -3dB) to assure accurate response to low-frequency soundtrack information
- **Sensitivity** (>89 dB SPL) to insure adequate efficiency

- **Continuous power handling** (100 dB SPL in 3000 ft^3 room for three hours) to guarantee reliability and sufficient dynamic headroom. (This specification applies to one or two subwoofers as a manufacturer may elect a design approach that results in two totally separate enclosures but sold as a single system.)

THX also requires that powered subwoofers meet certain electrical specifications so that they perform identically to passive subwoofers driven by THX-certified amplifiers.

SOME MISCELLANEOUS DETAILS

Before we end this Section, there are a few additional points you should be aware of.

System Calibration

After you install and hook up a home theater system, the important next step is "calibrating" the system. This simply means that you use the internal test tone generator included in every surround processor and adjust each speaker's output so that it is equally loud when heard at the viewing/listening position.

You can "earstimate" the levels and do a very satisfactory job or you can opt for a relatively inexpensive sound pressure meter and use its readings as a guide. In either case, follow the calibration instructions included with your surround processor.

Calibration's goal is simple: The creation of a single coherent soundfield between front and surround speakers. Most people have a tendency to turn up surround gain until they hear the surround speakers as a separate source. Generally speaking, this is **not** good.

Surrounded By More Than Is Good For You?

When watching movies, surrounds that are too loud draw attention away from the screen and destroy the sense of envelopment. With music, too-loud surrounds will amplify the ambient or other rear-channel effects and destroy the balance carefully created by the producer and engineers.

In short, improper surround balance may be pleasing but it isn't accurate. Although we encourage you to tweak levels using program material, avoid the temptation to turn the surround up until they are as loud as the front speakers. Remember that directors and sound engineers usually intentionally mix surround material so that it is lower in level than front channel information.

Bass-Heavy Systems

Another ill that proper calibration can help cure is the all-too-common symptom of heavy bass. Although we're well aware of the emotional weight and impact clear and solid bass reproduction adds to any source, we're also aware that most folks (salespeople and consumers alike) seem to like more bass in a home theater system than they'd ever find in the real world.

OK, so turning up the LFE channel is easy. It might impress your uncle or distress your neighbor (either of which might be temporarily desirable) but, in the long run, it's far from natural and is really antithetical to the idea of good home theater or high fidelity. A subwoofer should be totally inaudible when there are no prowling dinosaurs in the soundtrack or pedal notes in the organ symphony you're playing. If you hear a heaviness surrounding everything you play through a system, the chances are good that the bass is turned up too loud.

Perform a system calibration again. Note where the levels should be set. Leave them there and let the director or musician determine how much bass you should hear.

Oh, well, you may really want to "muck around" anyway. Do it. Enjoy it. Maybe you'll even leave the bass output a few dB above what the calibration tone says it should be. Just remember that you're not necessarily hearing things the way that sound's creators intended you to.

Hum

One of the most irritating distractions in a home theater system is "hum." (And no jokes about not knowing the words, OK?) If hum persists after you've checked all the usual suspects (cables, AC wiring, AC polarity, etc.) **and** the home theater system includes cable, we suggest you isolate the cable downlead from the rest of the system by using two 75 – 300 ohm matching transformers in a back-to-back configuration. Unscrew the cable itself from the converter box's input, install the transformers and reconnect the cable to the box. This should eliminate the effects (hum is the audible one!) of any difference in ground potential between the home theater system and the cable network. By the way, these transformers are easy to find at any electronic parts store and are very inexpensive.

Further Reading

The science of acoustics is so complex that any short treatment is bound to be incomplete. Although we're sure you'll find the information in this section accurate and useful, you may want to read a bit more.

If so, the best book we know of is F. Alton Everest's *The Master Handbook of Acoustics (4th Edition),* published by Tab/McGraw-Hill. It is thorough and complete but delightfully free of abstract mathematics. Mr. Everest's writing style is direct and to the point and his command of the subject is superb. We recommend it highly.

THE PRACTICAL BASICS OF PLANNING, SELLING, AND BUYING A HOME THEATER SYSTEM

PREVIEW

This Section gets down to the nitty-gritty of intelligently selling a home theater system. **Frankly, it's written primarily for the retail sales professional** but consumers will find the information useful, too.

We'll review various approaches (adding-on to existing systems as well as "start from scratch" systems) and how to present them.

If you're a salesperson, you should read this section to refine your knowledge, and to make sure you and your demo facilities are "up to snuff." A well-orchestrated demonstration will do more to help you establish yourself as a trusted professional than almost anything else you can do. So pay attention. Go through "the drill" and you'll sell more.

Are you a customer? Then you should know a bit more about the salesperson's world. Know what to expect from a demo. Know how to distinguish a good one from a mediocre one. Most of all, know that you should buy from the person who answers your questions most clearly, who presents a system that truly fulfills your home entertainment needs.

In short, this Section will help everyone who wants to sell or buy a home theater system. And that pretty much includes anyone who's reading this book, doesn't it?

GETTING THE BASICS RIGHT

As we've emphasized before, a home theater purchase is a lifestyle purchase rather than an equipment purchase. But that doesn't take away from its complexity, particularly from the consumer's point of view. In order to simplify the choices, let's take a look at the various home theater configurations we're likely to run into. As we do, remember the advantages inherent in "stair-stepping" your customers through this same process. Home theater doesn't have to be an "all at once" purchase. For most customers, in fact, home theater systems are amalgams of components combined over several years as people get more comfortable with the serendipitous advantages of having both audio and video components work together.

WHAT DOES YOUR CUSTOMER REALLY WANT?

Let's look at the basics. You and your customer face a real choice when you start talking about "home theater." First, there may be a real difference between what you mean by that term and what your customer means.

Before you even begin a sales presentation, you have to determine what your customer wants. This doesn't necessarily imply that your customer really knows. After all, the choices are enormous and it's part of your job to make some initial sense out of the seemingly endless choices.

THE CHOICES

Does your customer want a full-scale dedicated theater room with custom seating, lights that dim automatically when someone presses "Play," curtains that sweep into place to block distracting sunlight, an extensive loudspeaker array, and the latest video projector?

Or is it a generic HTIB ("home theater in a box") system that your customer can hook up in an hour or so. Even if this answer isn't the most exciting – or potentially profitable – one for you, remember that it's a lot better than a VCR connected to a TV, isn't it?

Unless you're working at a very upscale store, most of your customers are probably aiming at something between these solutions. After all, a dedicated room is a luxury few have. And the compromised performance typical of most all-in-one systems simply isn't that exciting, even to relatively unsophisticated customers.

ARE COMPONENTS THE ANSWER?

That leaves a *component* system – one composed of carefully chosen separate yet complementary pieces like a DVD player, amplifier, surround processor, etc. – as a very good choice. With a component system, you can offer the performance typical of the finest custom-installed theaters. And the operating convenience that is supposed to be – but often isn't – the hallmark of the all-in-one approach. In addition, separate components offer significant advantages in flexibility as well as an easier upgrade path when the time comes (as it eventually will) for your customers to improve their systems even more.

SOME PRACTICAL TIPS

One of the first questions a salesperson needs to ask when you're beginning a home theater presentation is "Do you have any equipment now and do you want to use it as part of the new system you're thinking about?"

Let's look at the possible answers to this question and what they mean for the presentation you're about to make.

If the answer is "We don't have anything right now — what do we need?" you have a clear run at presenting what you think will best serve your customer's interests. Of course, you still have to determine real needs and effective ways of meeting them but at least you don't have to balance those needs with the capabilities of previously purchased and possibly inappropriate equipment.

If the answer is "We have a good stereo system at home and I'd like to use as much of it as I can," you are well on the way to presenting add-on components to enhance the capabilities of what's already there.

If the answer is "We have a good stereo system already but we're not sure what we want to do with it," you have a potentially rewarding challenge ahead of you. This situation may evolve into a magnificent selling opportunity or a confusing morass that'll turn a potential buyer into a "Be-back." (Be-backs are so named because they usually say something like "I want to think this over and I'll *be back* tomorrow.") Of course, you'll rarely see these customers again. Something you said – or *didn't* say – put them off. Be-backs are generally created by confusing or ineffective presentations. If you're getting more than your share, look seriously at your own selling skills — they made need a bit of a tune-up.

TYPES OF HOME THEATER SYSTEMS

The "Lonely TV" Theater

As we've already pointed out, the simplest "home theater" is the classic "TV, VCR or DVD player, and some popcorn" combination. Although not home theater in any real sense, it does provide enjoyment for many.

TV and A Stereo System

The next step usually involves audio equipment. Here, the TV/VCR/DVD player combination (and cable/satellite box if present) connects through a stereo receiver or preamp/power amp combination to a pair of loudspeakers.

The advantages here are enormous. In addition to getting true stereo (almost all VCRs today have stereo-capable tuners to decode over-the-air and cable broadcasts and all DVD players pass a high-quality stereo signal through their analog outputs), the more powerful amplifier in a receiver and the broader bandwidth speakers attached to it will produce much more convincing sound for a larger number of people. There'll be more bass and more high-frequency details than you'll ever hear through TV speakers alone. And the cost of creating a home theater system if your customer already has the video gear and an audio system? Maybe $25 or $30 depending on exactly what cables are needed to complete the hookup. That's just a bit more than a pair of first-run movie tickets!

Although some customers may be astounded by the improvement this hookup provides, there are several drawbacks:

1) **No center channel speaker**: The sound won't be anchored to the screen as it would be if a Dolby Digital or DTS decoder were processing a multi-channel soundtrack. That means fewer people will benefit from the precise aural imaging that's an intrinsic part of good home theater.
2) **No surround information**: Your customers won't enjoy the full impact and involvement that a correctly reproduced surround signal can generate. Although there are relatively inexpensive devices that simulate surround, their effect is variable and, in most cases, little more than accidental. In short, it is

about as similar to what the director wanted you to hear as a Yugo (remember the Yugo?) is to one of today's better cars.

Moving To The Surround Experience

The next step, from stereo to full digital processing, also represents a substantial improvement.

From the audio perspective, your customer can make this jump by choosing to:

1) **Add a multi-channel preamp/processor, three additional channels of amplification, and three speakers** (center channel and two surrounds) to the existing system. Although this allows your customer to retain existing speakers (they become the main L & R speakers in the new multi-channel system) and the power amp (if it's a full separates system) or the power amp section of a receiver or integrated amp (*if* that component has the necessary jacks), this isn't a particularly good approach.

   It leaves you with the task of helping your customer choose a center channel speaker that matches those older (possibly *much* older) speakers acoustically. That may be difficult, particularly if your customer is very sensitive to tonal differences. (Adding surround speakers isn't as problematic.) And, of course, the choices of preamp/processor and additional amplification are extensive.

   As you are beginning to see, the major disadvantage here is that this tactic isn't as economical as it might first appear. It's a difficult acoustic challenge and often presents unwanted wiring complexity as well as component-to-component compatibility issues. We strongly suggest you go to …

2) **Use the existing system in another location and buy a new home theater system.** . Some customers might see this as a bit self-serving. "But we already have this good stuff," they might be thinking. True, their old equipment may be good in that it works well and sounds just fine but, at best, a cobbled-together multi-channel surround system based on a stereo rig is going to be difficult for many family members to enjoy.

   Instead, we suggest that you present the idea of using that perfectly good system elsewhere – another bedroom, a vacation house, possibly an office. Although most salespeople see this as the more desirable approach, it usually carries the implied resistance of a higher (but not *that* much higher) purchase price. Still, this needs to be handled in a thoroughly professional manner. This "start from scratch" approach opens the door to a system design unencumbered by older components and leaves you much freer to then examine your customer's needs in a much clearer and less-confining way.

THE NUTS AND BOLTS OF PLANNING: WHAT YOU NEED TO KNOW

With the basics out of the way, your next job is to make sure you understand your customer's real needs and limitations. Here are some of the questions you need answered in order to prepare the proposal that will get the sale.

Will the home theater room be "dedicated" (used only for watching or listening) or will it be "multi-purpose" (a den, family room, etc.)?
If "dedicated," ask the following:

- What is the relative location of this room to other living spaces?
- Will sound leakage be a problem? (Blasting the latest action flick while the baby's asleep in the next room will probably raise both the baby and serious restrictions against ever doing THAT again.) Remember that you might need some kind of structural analysis to determine the degree of acoustical isolation.
- Will you need to add a layer of sheetrock to the walls or do some other work to isolate the home theater room?
- Will there be more than one row of seats?

  If "multi-purpose" (and most sales fit into this category) you'll still need to know the answers to most of the above. The only difference (at least at this stage of the planning process) between a dedicated and a multi-purpose home theater room is that a multi-purpose room won't have several rows of seating.

  In addition, you need to know all the other uses the room will be put to in order to estimate traffic flow, obstructions, etc.

  In **all** cases, you'll need to know the kind of electrical service available in that room. **Don't underestimate the importance of this one!** Survey the room to determine if there are separate 20 or 30A circuits for high current amplifiers, video projectors, etc. and where all AC outlets are located. You may need an electrician to run new lines or reroute existing ones so plan carefully here.

Is an interior decorator or architect involved in any way?

- If "No," breathe a little easier as your job just took a welcome downturn on the Cosmic Difficulty Scale.

- If "Yes," make immediate plans to meet with these folks as soon as possible.

The more quickly they begin to feel confidence in your professional approach, the more easily you'll complete the sale. (You will rarely, repeat *rarely*, win any battles here so don't even start. Cooperation and patient explanations on your part will carry the day – if the day can be carried!)

What are the potential listening and viewing angles in this room?

This can be a complicated step. You need fairly accurate dimensions and detailed knowledge of windows, doors and other breaks in the wall space. If your customer is an audiophile, you'll need to put even more weight on that side of the planning process (room mode analysis and suggestions for correcting acoustical anomalies, etc.)

To better plan the video side of the system, you need to help your customer select the video device (direct view, rear projection, front projection, flat screen display) for this room. Here are some considerations:

- For direct view sets:
    - Comparatively small size makes placement easier
    - Relative immunity to ambient light means your customer might get other uses out of room while system is in use.
- For projector sets:
    - Large size (or two-unit design) makes placement more difficult.
    - Projector sets are much more affected by ambient light. Don't place a projector set, for example, where the screen will be hit by direct light. If you're using a front projector, make sure the area immediately behind the screen is DARK — the darker, the better. If the system is to be used during the day, make sure that the entire room can be darkened sufficiently so that the image isn't washed out.
- For flat screen displays
    - Mounting questions, signal routing, etc.

How Big Is the Image Going To Be?
Where Will The Loudspeakers Go?
Where's The Main Viewing/Listening Location?

These are all interdependent factors and the final plan will almost certainly be a compromise. If politics is "the art of the possible," so is designing a real-world home theater system. They're both successfully practiced only when you've artfully balanced the often conflicting requirements of audio and video factors. For example, your customer may want a front projector. That's fine but how do you isolate it so noise isn't a problem? You may need to install this equipment in a sound-reducing housing or baffle. Audio factors might include room EQ, substantial acoustical treatments, etc.

THE DEMO

If success in real estate is controlled by the three rules "Location," "Location," and "Location," selling audio and audio-video components is equally governed by "Demonstration," "Demonstration," and "Demonstration." Make no mistake about it, the "demo" is a critical element in a successful sales presentation. Contrary to popular rumor, demos aren't easy to set up and they aren't always easy to control. But they're essential. Here are some thoughts on techniques we've found useful over the years:

Demos Have A Purpose!

Demos should present ***benefits*** – not technology. A system properly set up and demoed will leave your customers saying little more than "Wow!" And that's the point.

Demonstrations should appeal to the emotions, not the intellect. You can explain technology (carefully, mind you) before the demo. You can reinforce a technical point after the demo. But during the demo, let the system do the work! Remember that the vast majority of today's customers aren't impressed with knobs and buttons. They simply want to enjoy a symphony, a rock recording, or a great movie.

So, if they comment about the complex system you've just shown them, you may well have blown it! If they mutter about the material you used, you missed! If they emerge holding their ears because you couldn't resist the urge to "turn it up" just a bit more, you really goofed!

The All-Important Pre-Demo System Check

Anyone who starts a demo without knowing what's connected to what and how things work ***in that system*** is asking for trouble. There's nothing worse than taking a customer into a demo room only to fumble behind a rack of equipment trying to figure out why you have Lenny Bernstein conducting the Vienna Philharmonic on screen and Black Sabbath coming through the speakers!

Remember that most customers are intimidated enough with the concept of a home entertainment system. If they see you – the supposed expert – momentarily stumped, they'll probably just murmur politely as they head to the door. Avoid this by checking your primary demo system ***daily***! Make sure everything's still hooked up the way you remembered and that all the default settings on a surround processor, for example, are correct. You never know when you'll need quick access to a properly functioning system to clinch that important sale.

Setting The Stage

When you demo isn't as important as how you set the stage. And "setting the stage" is just the right phrase, too. A demo is a performance in every sense of the word. The more drama, the more anticipation you create (providing that the demo fulfills those expectations, of course!), the better your chances for a successful sale!

> *Although effective demos sometimes occur early in a sales presentation, most experienced salespeople prefer to use them as the final tool to close a sale. If you follow conventional wisdom and place the demo towards the end of your effort, be very careful that it goes flawlessly. Nothing undermines your position – or negates all the work you've already put into the presentation – more than an ineffective demo.*

Before you begin the demo, focus your customers' attention on what you want them to see or hear. In other words, tell them what to expect. You might mention some subtlety in a soundtrack, some image shifts that might otherwise escape their notice, etc.

Here's an example:
Salesperson: "Maybe this is a good time to take a look and listen to what we've been talking about — I think you'll find this really interesting.

"I'm going to play a short scene from *Toy Story*, a really great animated movie with a wonderful soundtrack. You'll hear lots of different things — music, really clear dialog, even some pretty subtle crickets in the background as the two main characters stand all alone in a gas station. And then a truck pulls in and, well, you'll probably think you're about to get run over! Here it goes …"

Notice what you've done:

1) You've told your customer that you're not going to hold him for ransom with a lo-o-o-o-ng presentation.
2) You've created the positive anticipation of seeing a good movie clip.
3) You've explained some of the key elements you want the customer to listen for.
4) You've promised some visceral excitement ("you're about to get run over").

In short, you've told the customer exactly what to expect. They're prepared. And they'll be waiting to see and hear what you've promised!

Orchestrating The Demo

You've got two options as you demo a system.

1) The "You Do It" Demo:

You control the demo yourself (sometimes a ***very*** good idea, incidentally!) by bringing your customer to the display area or demo room and running through the presentation relatively quickly yet enthusiastically. This style works well when the sales floor is busy or when you judge a customer is not yet a serious prospect. If you choose this approach, here are some guidelines:

a) **Keep your demos short.**

b) If you use more than one selection, **set the stage for each one separately**.

c) **Pre-select your demo material** (one selection for a family presentation, one "testosterone special" with plenty of special effects, a classical music video, etc.) ***and note effective passages*** (chapter or timing information) ***on the cover***! Do this with **every piece** of demo material you use. ***Remember, even if you've seen the same scene or listened to the same track 10 septillion times before, your customers have not! And they're the ones who need to be impressed.***

d) **Use appropriate material**. Don't send the five-year-old kid out of the demo with nightmares for the next six months after seeing Arnold rip the face off yet another alien! You will *not* get referrals this way.

e) **Leave your customers wanting more**! Don't play the whole movie or album, just enough of it to get your points across clearly and effectively.

f) **Emphasize the performance capability and the value** of the equipment that just raised their goosebumps!

At the end of the demo, your customers should want to see and hear more. But that should happen in their own homes! After all, that's the point of the whole thing, isn't it?

2)The "Interactive Approach"

This gets your customer involved in operating the equipment and is usually a better bet when the sales floor is less frantic or when you've got a serious prospect you want to spend serious time with. For this demo, here are some suggestions:

Get your customers involved quickly. Give them the remote. Tell them what buttons to push to change inputs, raise and lower volume, etc. **Have them do it**!

This gives your customers a strong sensation of control. Once they feel comfortable with the equipment, the thoughts of actually owning it are just a hop, skip, and a jump away!

Follow all the steps above. Use short selections of appropriate demo material, frame each selection by telling your customers what to expect, leave them wanting more, and highlight performance and value!

Following Up

When the demo is over, give your customers a few seconds to think. This reinforces the impression you've created and is a great time to gently remind them of the performance and value inherent in the system you've just presented. Tell them again what they've seen and heard.

This is sometimes called the "*Tell 'em what you're gonna tell 'em, tell 'em, and then tell 'em what you've told 'em*" technique. It works! Why? Because it concentrates on *benefits*! Notice that you haven't talked about technology at all. You haven't confused them. You haven't left yourself open for a question you can't answer. You've simply shown your customers what the technology in that mysterious black box will do for them! That's a powerful tool that you need to use time and time again as you present systems to the folks who walk into your store.

No, we're not downgrading the value of technical knowledge. (We wouldn't have written this book if we believed that technology isn't sometimes interesting for its own sake.) However, we know very well that most customers don't want to know a lot about the technology. They want to *use* and *enjoy* it. That's all. And that's enough.

In Summary

Demos appeal to the emotions and you'll need to judge where in your presentation a demo will be most helpful. The important points are:

1) Telling your customers what to expect
2) Showing them what you've already told them to expect
3) Telling them what you've just shown them.

Remember that the most important element of any demo is ***excitement!*** That's what really sells the experience. Potential customers come in your door because you can show them the difference! If you don't, nothing separates you from "Mega-Buy Electronics," your mass-merchant competitor down the street. Remember they have those nationally advertised brands and would surely love to compete with you solely on price! Your ace is your ability to present product value and your expertise. What better way to do that than with goosebumps?!

Some Questions We've Heard . . .

OK, we're going to talk a bit about selling Rotel here. Ah, yes, crass commercialism rears its ugly head again! But the fact remains that some of your customers will have questions, even after a really effective demo and some of those questions will be about Rotel.

Even though you now have far more information about the company than you had when you started reading this book, you might find the following helpful. Yes, we know that no sales presentation ever follows a script, but you might get some good ideas anyway!

Customer: Wow, this is really good but it's a bit more expensive than some of the advertised names that I know.
Salesperson: Actually, Rotel is a great entertainment investment. It's well-built and provides exceptional sound quality for a relatively small outlay. Think of it this way — you're probably going to own your system for years and a few dollars more for something you'll enjoy for those years is really a smart move. And it's expandable: You can add system capabilities as your needs change but still enjoy the same quality.

Customer: Well, I've never heard of the name. How do I know they'll still be around five years from now?
Salesperson: Obviously, no one can guarantee where they'll be in five years! But Rotel has been in the audio business for over 40 years. It's a reliable, family-owned company specializing in home theater and audio components. Because the company concentrates its efforts, their engineers really know their stuff.

Customer: Still, how do I know they make reliable products?
Salesperson: Well, to begin with, they manufacture their own products in their own factories! That gives them the advantage of total in-house control. As an example, Rotel actually makes the power supply transformers that act as the backbone of a component's performance. That's really different from some companies who design and then "contract out" their products and have no real control of how their units are put together. And, because Rotel makes their products better, they back them better. Rotel has one of the best warranties in the business: Five years on electronics and two full years on mechanical things like DVD and CD players. (U.S. warranty, other countries many vary.)

Customer: I'm sorry, I'm just not going to buy something I've never heard of.
Salesperson: I understand your concern. However, some of the best ears in our industry, credible reviewers who make their living evaluating products and reporting their findings, have written reams of paper on Rotel's exceptional value. And the company has earned quite a number of international awards over the years for exceptional performance and value. Here, let me show you some highlights . . . (Let's go to the reviews!)

Will these scenarios ever happen in your store? Probably not – and certainly not as we've written them here. But they do point to several ways of positioning Rotel with those customers who don't read all the home theater and high fidelity magazines and who consequently don't know Rotel's long tradition of excellence.

PREVIEW

Successful custom installations result from combining several sets of skills guided by extensive knowledge of acoustics, construction techniques, engineering, and a dash of elementary physics. Oh, yes, there are a plethora of other disciplines, too. If you're getting the idea that this Guide is *not* going to turn you into a master installer overnight, you're right on the money!

But what you *will* learn here are some of the things that go into a successful installation and what you can do as a consumer and as a salesperson to make sure everyone is satisfied when you fire the system up and settle back to enjoy your favorite source – be it a symphony, a movie, or a playoff game in vibrant color and bleacher-rattling surround sound.

Simply because an installer must draw on so many different skills, it is impossible to include them all in this Guide.

A quick list of areas you need to know something about might look like this:

Acoustics; room design and isolation; equalization and room treatments; measurement techniques; audio and video signal processing and distribution; TV/projector calibration; video image enhancers (line doublers, quadruplers, interpolators, etc.); screen types, applications, and mounting techniques; shielding techniques against RFI (radio frequency interference) and EMI (electromagnetic interference); IR (infrared) technology; voice and data transmission; Internet access; security; wire types and applications; HVAC (heating, ventilation, and air conditioning systems); lighting systems design, installation, and controls; systems integration; blueprint reading; general contracting; and basic electrical engineering (AC and DC circuits).

Quite a bundle of knowledge, isn't it?

In addition, the successful custom installer has a full set of business skills to master or there simply won't be a business for long.

ROTEL'S CONTRIBUTION

So how can Rotel help? More specifically, how can this Guide help? Perhaps we can be most useful by keeping our comments here as short as possible. We *can* suggest that consumers think about what they want in a custom installed-system carefully before visiting your first installer. Take the time to map out everything you want such a system to do. The installer you finally choose to help you achieve those goals will work with you to refine that list but you have to start the process yourself.

The best recommendation we can give our dealer salespeople is to ***listen*** to the consumers who come into your stores. You may know the technology better than they do but they know their needs better than you. The truly successful installation comes from pooling that knowledge.

And we'd like to remind all of you that Rotel's "custom-install" products are built from a different point of view: *We think good sound comes first.* That's why *all* of our products (audio, audio/video, and custom installation-specific) come from the same award-winning international engineering team. We use the same Balanced Design philosophy, the same care in selecting appropriate parts, the same painstaking assembly and test procedures, as we do with everything we make.

Are our goals exactly the same for all products and all applications? Yes. And no. We realize that while an audiophile may tolerate a more finicky amplifier, and even endow such a product with a "personality" that's part of some mystique, a custom installer's patience for these things is less than zero. As much as we push ourselves to produce *reliable* products for all our customers, we're even more conservative about custom-install products that we envision being "buried" in a closet, an equipment rack, or on a high shelf.

Another area we're particularly demanding of ourselves in is *easy operation*. We know that owners of custom-designed systems are far less interested in technology that they are in technology's benefits. That's why we've gone to such lengths to make our custom-install products so easy to use.

As a final note, remember that Rotel designs entire systems so we're very conscious of *how things work together*. That includes system simplicity and protecting signal integrity over the long cable runs typical of today's multi-zone systems.

So, as you review your needs for a custom system, we would appreciate your remembering Rotel.

There is one organization which can help both consumers and dealer salespeople reach their goals for a custom system – CEDIA (the Custom Electronics Design and Installation Association).

Consumers can use CEDIA's web site to find a member dealer. Dealer salespeople and installers can use CEDIA's extensive education programs to keep up with the latest technologies and techniques.

You can contact CEDIA at:

CEDIA

9202 North Meridian Street, Suite 200

Indianapolis, IN 46260-1810

(800) 669-5329

(317) 571-5603 (FAX)

www.cedia.org

AC-3 (now called Dolby Digital)

A method developed by Dolby Laboratories for "compressing" (or reducing) the amount of bits needed to convey multi-channel, full bandwidth digital sound for a wide variety of formats including DVDs, laserdiscs, television and satellite broadcasts.

AC-3's potential benefits include fully discrete sound for five full bandwidth channels and one low frequency (subwoofer) channel as compared to earlier "matrix" formats with mono surround and a derived subwoofer channel. The "AC" stands for "Audio Coding" while the "-3" identifies this method as the third one developed by Dolby for this purpose. (See also: Dolby Digital and Dolby Stereo Digital.)

"Academy" filter

A high-frequency filter used in some advanced home theater controllers to compensate for the overly bright soundtrack of many earlier films. This filter operates only in the Mono mode and begins attenuating high frequencies just below 2 kHz. It reaches maximum effect (-16 dB) at 10 kHz. Note that the "Academy" filter is NOT the same as "Re-equalization," a technique unique to THX processing, in which high- frequency content to ALL front channels (LCR) is reduced less radically.

Ambience

The surrounding "atmosphere" that frames more intense sounds. In audio reproduction, ambience means the subtle, very low-level sonic cues that describe the acoustic environment in which a recording was made.

In video, these very faint sounds (wind or traffic noise) add to the reality of a scene. Often recorded separately from the dialog or main effects tracks, they are usually added later in the "post-production" phase of preparing a film for release. Sometimes called "Atmospherics."

Baffle

The flat or mildly curved front panel of a conventional loudspeaker enclosure that holds one or more active drivers. Some loudspeakers have drivers mounted on more than one surface. Technically, each surface on which a driver is mounted is a baffle but the term is usually reserved to the surface holding the most drivers.

Bipolar

A loudspeaker that radiates equal amounts of energy in two opposite directions with the same polarity. The compressions and rarefactions that constitute sound waves move outward from each side of the enclosure symmetrically: i.e., both sides of the enclosure produce a positive pulse or a negative pulse at the same time. (See also: Dipolar) Also refers to a particular type of transistor.

Bitstream Converter

A one-bit or low-bit A/D and D/A oversampling conversion method developed by Philips in which the audio signal is represented through PDM (Pulse Density Modulation) or time averaging at a frequency of 11.3 Mhz.

Bridged Mode

Some amplifiers have the ability to be operated in a "Bridged Mode" which inverts one channel of a stereo amplifier and places it in parallel with the other channel, in effect turning the amplifier into a mono unit. This increases the amplifier's output voltage which results in up to triple the rated output power available in stereo mode. Note: When operating an amplifier in "Bridged Mode" use only nominal 8-ohm loudspeakers.

CAV

Constant Angular Velocity. Laser disc operating format in which the disc rotates at a constant speed during play. CAV permits more special effects (still-frame, slow-motion and fast-motion for example) but this format is somewhat wasteful of disc space. CAV discs are limited to 30 minutes of material on each side of a 12" laser disc. (See also: CLV)

CD-ROM

(Compact Disc - Read-Only Memory): Refers to both the media and the format for storing digital data (computer files or music information) on Compact Discs. Because of its high information density, an ordinary CD-ROM can be used to store up to 680MB of digital information.

Clipping

A signal-altering condition usually occurring when an amplifier cannot accurately amplify the shape of the input signal because of current and voltage limitations. When viewed on an oscilloscope, a "clipped" signal shows substantial flattening of both positive and negative peaks — almost as if these peaks had been "clipped off" by a pair of shears. A clipped signal contains large amounts of harmonic distortion and can be more damaging to loudspeakers than an undistorted signal of equal or even greater power.

CLV

Constant Linear Velocity. Laser disc operating format wherein the rotational speed of the disc varies as the laser pickup travels from the inner edge to the outer edge of the disc. This maintains a constant velocity of data past the laser pickup and allows more efficient use of disc space resulting in 60 minutes of material on each side of a 12" laser disc. The downside is that CLV does not allow the special-effects capabilities of the CAV format. CLV is also the operating format for Compact Discs. (See also: CAV)

Compression

A way of reducing the storage requirements, transmission time, or both, of digital data. Compression can be either "lossy" (i.e., some data is irretrievably thrown out during the process) or lossless (the recovered data is a bit-for-bit replica of the original).

Continuous Calibration

A low-bit D/A converter with a constant weighing circuit designed by Philips to keep the D/A process linear and improve resolution.

Crossover

A circuit (technically, a superposition of at least two filters) that divides an audio signal into parts above and below a particular frequency called the "corner frequency" or "crossover frequency." The rate at which a crossover divides the spectrum is called the "slope." .

If you see a crossover described as "400 Hz @ 18 dB/octave," the crossover's corner frequency is 400 Hz and the low pass filter reduces high frequency output by 18 dB one octave *below* that point while the high pass filter reduces low frequency output by 18 dB one octave *above* that mark.

Crossovers are usually used to separate a wide range audio signal into narrower-range components so that each component may be safely directed to specialized loudspeaker drivers (woofers, tweeters, etc.) for optimum reproduction. Crossovers are usually categorized by their complexity (2 way, 3 way, 4 way, etc.)

Functionally, crossovers are categorized as "passive" or "active." Passive crossovers are sometimes called high-level crossovers and are usually found inside loudspeaker enclosures. They divide the audio signal after it has been amplified and send the different portions to the appropriate loudspeaker system drivers. Active crossovers (sometimes called electronic or line level crossovers) divide the signal prior to amplification and thus require separate amplifiers for each frequency segment. Active crossovers are more precise but usually more costly to implement correctly.

D/A Converter

The Digital to Analog converter is an electrical circuit which converts a binary coded word (a "sentence" of 0's and 1's) into an equivalent analog voltage (continuous waveform).

Damping

In an acoustical sense, we describe a room that doesn't reflect sound very well as one that's heavily "damped." These rooms usually have curtained walls, sound-absorbent ceilings, carpeted floors, lots of overstuffed furniture, or some combination of these "damping" elements. People also provide damping – an empty theater is more reflective than when it contains a large audience.

D'Appolito configuration

Named after its designer, Joseph D'Appolito, this is a way of restricting a loudspeaker's dispersion and limiting the area into which it radiates sound. Speakers using the D'Appolito configuration place a tweeter between two midrange drivers. The size of the drivers and their spacing is critical. When properly executed, a D'Appolito configuration results in comparatively little radiation above and below the center axis of the speaker.

Data Reduction

Also called Data Compression. Any technique that reduces the amount of digital data required to represent a given amount of information. Data reduction enables large amounts of information to be easily and efficiently stored and transmitted. For example, a technology that reduces data by 75% (4:1 compression) can store a 16-bit digital word in just 4 bits. Data reduction technologies used for digital audio must have very carefully designed parameters so as to have a minimal effect on the accuracy of the reproduced sound.

Delta/Sigma

A so-called "single-bit" digital signal format where each sample is described in relative terms only. In other words, a sample in a delta-sigma datastream can be only stronger or weaker than the preceding (or following) sample. Delta-sigma signals are inherently less data-intensive than their PCM equivalents but can't be mathematically manipulated as easily which makes them somewhat more difficult to process. (Sometimes called *sigma-delta*.)

Diaphragm

The surface of a driver that actually pushes against the air to generate sound waves. Examples include a cone or dome (dynamic speaker) or a membrane (planar speaker).

Digital Coaxial Output

An electrical output connection for the raw digital datastream. (See also: Digital Optical Output)

Digital Filter

Any filter used in the digital domain. Rotel CD players use oversampling to raise unwanted frequencies away from the audio range. Digital filtering does not produce the phase distortion common to analog filtering.

Digital Optical Output

An optical output connection for the digital datastream. Standard connection types are Toslink and AT&T. (See also: Digital Coaxial Output)

Dipolar

A loudspeaker that radiates equal amounts of energy in two opposite directions with opposite polarity. The compressions and rarefactions that constitute sound waves move outwards from each side of the enclosure in inverse symmetry: i.e., while one side of the enclosure produces a positive pulse, the opposite side produces a negative pulse. (See also: Bipolar)

Direct Circuit Paths

Rotel employs remote switching wherever possible. In this fashion, we bring the switch to the circuit, rather than lengthen the signal path. Contact points are also kept to a minimum. This helps us to prevent RF pickup and achieve a lower subjective noise floor.

Direct Radiator

A loudspeaker that radiates its energy in one primary direction.

Dolby Digital (AC-3)

This is Dolby's latest technology that utilizes a digital delivery system and 5.1 channels of audio. There are five discrete "full-bandwidth" channels (left, right, center, left-rear and right-rear) as opposed to Pro-Logic's filtered 100Hz -- 7kHz surround channels, plus a separate bass only effects channel (which because the frequency range occupies only one-tenth of the full audio spectrum was given the designation ".1").

Dolby Noise Reduction

Any of a number of double-ended (record encode/playback decode) systems designed to reduce the noise inherent in analog tape recording and mixing. The first example, Dolby "A," was designed for professional applications. Dolby "B" was specifically developed as a consumer format. Dolby "C" and "S" formats, both consumer technologies, expand B-Type's capabilities. Dolby "SR" is an advanced professional technology.

Dolby Pro-Logic

An advanced consumer version of commercial Dolby Stereo film sound matrix decoder technology. Compared to standard Dolby Surround decoding, Pro Logic adds center channel capability and enhanced separation thanks to special "steering" circuitry for more accurate spatial presentations. It decodes Dolby Surround-encoded video software with the same accuracy and fidelity as commercial decoders used in motion picture theaters. (See also: Dolby Surround)

Dolby 70mm Six-Track

A commercial film sound process that records six individual soundtrack channels (Left, Center, Right, Left Surround, Right Surround, Subwoofer) on magnetic stripes that have been "painted" onto the 70mm film print. The soundtracks are then read by magnetic tape heads on the projector and reproduced in the theater. Dolby 70mm Six-Track is a discrete analog format using no matrix encoding or decoding. Films made with it can be shown only in specially equipped theaters.

Although it offers improved audio performance compared to Dolby Stereo, it is rarely used today because of the expense involved in making prints.

Dolby Stereo

A commercial film sound process that allows four individual soundtrack channels (Left, Center, Right, Surround) to be recorded onto a 35mm film print using only two optical soundtracks, and then recovered when the film is projected. Dolby Stereo uses an analog active matrix encode/decode process and either Dolby A-type or Dolby SR-type Noise Reduction to achieve high fidelity surround sound in motion picture theaters. It is today's "standard" motion picture multi-channel soundtrack format.

Dolby Stereo Digital

Also called Dolby SR-D, Dolby Digital or Dolby DD. A commercial film sound process that allows six individual soundtrack channels (Left, Center, Right, Left Surround, Right Surround, LFE or Subwoofer) to be optically printed onto a 35mm film and reproduced in commercial movie theaters. The information is digitally encoded using a proprietary data reduction process called AC-3, and printed on the film strip in the area between the left-hand sprocket holes. The data is read optically when a film is projected, then decoded and reproduced in the theater. Offers considerably improved audio performance over Dolby Stereo.

Dolby Surround

Can be used in either of two ways (#2 mentioned here primarily for historical reasons):

1) Name given to Dolby Stereo soundtracks that have been transferred to home video formats (broadcast, VHS tape, laser disc, etc.). With the exception of the Dolby A or SR Noise Reduction, a Dolby Surround consumer soundtrack is identical to the Dolby Stereo theatrical sound-track, with the four original tracks remaining encoded onto two stereo tracks. This enables software with Dolby Surround-encoded soundtracks to be played-back on home video systems that have mono, stereo, Dolby Surround or Dolby Pro-Logic capability.

2) Name given to the passive decoding technology used in early consumer products that can extract only three (LRS) channels of a Dolby Surround soundtrack's four channels of information. Largely replaced in consumer equipment by Dolby Pro Logic decoding capabilities. (See also: Dolby Pro Logic)

Dolby Surround Digital

1) Name given to Dolby Stereo Digital soundtracks that have been transferred to home video formats. The soundtrack is identical to the Dolby Stereo Digital theatrical soundtrack, with the six original tracks remaining encoded onto a digital track on the software.

2) Name given to the digital decoding technology that can extract all six channels of a Dolby Surround Digital soundtrack from a laser disc or satellite/cable transmission and reproduce them in home video applications.

Driver

In a loudspeaker, any element (dynamic woofers and tweeters, electrostatic panels, ribbons, etc.) that moves in direct response to a signal from an amplifier.

DSD (Direct Stream Digital)

A single-bit digital data structure used for high-density recording and SACD discs. DSD signals are composed of a very high-speed series of individual bits that define amplitude *differences* rather than absolute values.

DTS

Abbreviation for Digital Theater Systems, DTS stands for a number of surround sound formats that promise enhanced sound quality.

Dual Mono Power Amplifier

A dual mono power amplifier is essentially two separate amplifiers in a single chassis. This design achieves better separation between channels resulting in superior stereo imaging and greater dynamic control. The most expensive examples incorporate separate transformers for each channel. However, many models employ a high efficiency toroidal transformer with dual windings (one for each channel) with the rest of the circuitry for each channel being totally isolated.

Dubbing Stage

A specially designed room resembling a combined small movie theater and recording studio control room, used by film directors & sound engineers to create film soundtracks. A dubbing stage's audio characteristics are defined by SMPTE standard # 202 and match those of SMPTE-standardized movie theaters so that filmmakers will be able to accurately hear how their films will sound when shown to the public.

DVD Forum

An industry group of over 40 manufacturers and technology providers that oversees development and implementation of the technical standards for DVD hardware and software.

Dynamic speaker

Any loudspeaker that uses dynamic drivers. (See also **Moving Coil**)

Dynamic range

The difference, expressed in dB, between the softest and loudest sounds a particular component, system or medium can process without adding excessive distortion of having the signal "buried" in a residual noise floor. Related to "Signal to Noise ratio."

Effects

Literally speaking, any special effect other than dialog and music that occurs in a film's soundtrack.

Efficiency

A measure of the percentage of electrical input that a loudspeaker converts into acoustic output. Most conventional home audio speakers are roughly 1-3% efficient. (See also: Sensitivity)

Encoding/Decoding

To convert information from one form into another, and then back into its original form. All digital audio technologies use some form of encoding to store and transmit audio information as number codes, and decoding to convert the number codes back into audio information.

Envelopment

The characteristic of a listening space where the listener is surrounded ("enveloped") by sound. Proper surround-channel envelopment of the audience is considered a very important characteristic of high-quality home and commercial theaters.

Excursion

A measurement of how far a driver diaphragm can move. Look for "peak to peak" measurements. Some dynamic woofer diaphragms move more than a ½" when they're really pumping out the bass.

Gold-plated RCA connectors

Improved connections minimize signal loss and resist corrosion for long-term reliability.

"Foley" effect

A subcategory of "Effect" that usually deals with smaller sounds like footsteps, door closings, etc. Usually recorded separately on a "Foley" stage equipped with many different devices for producing sound effects of various kinds. .

Insertion Loss

A drop in signal level caused by the addition of a component to the audio signal path.

Jitter

Minute variations in the timing of digital signals caused by a digital circuit's inability to "lock on" to an incoming data stream. D/A converters are particularly susceptible to jitter-induced problems. In audible terms, jitter causes a loss of transparency.

LCRS

Abbreviation of Left, Center, Right, Surround. The standard channel and speaker configuration for commercial and home theater sound systems. As the name implies, the L speaker is located to the left side of the screen, the C speaker is located at, or close to, the center of the screen and the R speaker is located to the right side of the screen. The Surround channel is reproduced in commercial movie theaters by multiple speakers located above, to the sides and behind the audience, and in home theaters by two speakers, usually located above and either behind or to the sides of the audience.

Line-Level

The relatively low signal level generated by most audio and audio/video source components, separate processors, etc. Output voltage for "line level" signals is typically in the 0.5 - 5-volt range, depending on what component is being measured.

Magnetic Soundtrack

Developed in the early '50s as a sonic improvement over older optical techniques, magnetic soundtracks are recorded on strips of iron oxide material similar to standard recording tape that are applied to the film after the photographic image has been printed. A projector read the soundtrack with a separate magnetic head just as a tape playback device does. Although early magnetic standards included provision for multiple discrete tracks and paved the way for further developments in surround sound, magnetic technology's cost and fragility made it vulnerable to new developments in optical technology, particularly Dolby Stereo. (See also: Dolby Stereo & Optical Soundtrack)

Magnetic Shielding

Techniques that prevents a speaker magnet's stray magnetic flux from interfering with the picture of a direct-view TV set, particularly when that speaker is placed very close to the TV set. A speaker designer shields a speaker by either specifying a magnet totally encased in mu-metal, a very expensive alloy with exceptional resistance to magnetic fields, or by fixing a second, flux-canceling magnet to the rear of the primary magnet. The latter approach is far more common.

For home theater applications, the center channel speaker MUST be shielded. Flanking (left and right) front speakers may need shielding if they're placed close (within 3') to a direct view monitor.

Mastering studio

The final link between a recording session and mass duplication of CDs. Mastering engineers use carefully constructed listening facilities to critically evaluate musical material and suggest often-subtle equalization and level changes, song sequences, etc., to make an album more coherent or convincing.

Matrix(ing)

The process of mixing two distinct signals with specific phase and amplitude relationships to form one signal so that the original components of the total signal can be separated at a later time. For example, the Dolby Stereo Motion Picture Sound Matrix mixes 4 independent soundtrack elements (LCRS) into 2 independent signals (Lt & Rt on the film) in such a way that the 4 original LCRS signals can be recovered and reproduced independently in a movie theater.

Metal-Film Resistors

Used for their low noise superiority, Metal-Film resistors feature a metal alloy which is deposited on a substrate such as plastic or ceramic. They are also available in very close tolerance specifications.

Moving Coil

Can indicate either a type of loudspeaker or phono cartridge: In this case, the term refers to a type of speaker driver that has a "motor" made up of a voice coil located in the gap of a permanent magnet. The coil, connected (either directly of through a **crossover network**) to the amplifier's output, moves in and out. Because the coil is mounted to the driver's surface (usually a cone or dome), it pushes or pulls that surface to compress or rarefy air molecules and thus create sound waves. Also called a **dynamic** driver.

Multi-bit Converters

Often described as 16, 18 or 20 bit designs. See the Digital Addendum for a full explanation.

Multi-Track

Two or more independently processed audio signals that are synchronized in time and intended to be heard simultaneously.

NTSC

National Television Systems Committee: The body that establishes standards for television broadcast and reception in America and Japan, (as well as a few other places). NTSC is also used to describe any hardware or software that is designed to NTSC standards. Sometimes jokingly called "Never The Same Color" due to technical difficulties occasionally encountered in accurately broadcasting or receiving NTSC signals.

Noise Reduction

Any technology designed to reduce steady-state noise (usually analog tape hiss) in the audio reproduction chain. Dolby Laboratories developed the first commercially successful professional noise reduction system (Dolby A-type) in the mid-1960s.

Octave

A range of frequencies defined by doubling – or, less frequently, halving – a particular frequency. The so-called "10 audible octaves" cover the normal human hearing range from 20 Hz to 20 kHz in the following way. Note that we've added a description of the musical tones they encompass.

1	*20 Hz to 40 Hz*	*Low bass*
2	*40 Hz to 80 Hz*	*Mid bass*
3	*80 Hz to 160 Hz*	*Upper bass*
4	*160 Hz to 320 Hz*	*Lower midrange*
5	*320 Hz to 640 Hz*	*Midrange*
6	*640 Hz to 1,280 Hz (1.28 kHz)*	*Midrange*
7	*1,280 Hz to 2,560 Hz (1.28 kHz – 2.56 kHz)*	*Upper midrange*
8	*2,560 Hz to 5,120 Hz (2.56 kHz – 5.12 kHz)*	*Lower treble*
9	*5,120 Hz to 10,240 Hz (5.12 kHz – 10.24 kHz)*	*Mid treble*
10	*10,240 Hz to 20,480 Hz (10.24 kHz – 20.48 kHz)*	*Upper treble*

Optical Soundtrack

A soundtrack printed directly on film stock exactly as image frames are printed. It appears as a continuously transparent strip of varying width against an opaque background. The projector reads the soundtrack with a photocell placed opposite a light as the film passes between them. Economically attractive, optical soundtracks first became popular in the early '30s. (See also: Magnetic Soundtrack)

Pan
In this case, a "pan" refers a sound that moves from one speaker to another. Examples include the sound of a car as it races across the screen from left to right or a helicopter that suddenly flies into the center of the action from behind you.

PCM
(Pulse Code Modulation) Often called "multi-bit." A digital data structure composed of "words" made up of a (generally even) number of individual bits, either "0s" or "1s" that completely defines the amplitude or strength of the sample it represents. Words follow each other at a very high rate called the "sampling frequency." For example, the PCM data structure used by the CD is often referred to as "16/44.1." This means that each word is composed of 16 bits and that words follow each other at the rate of 44,100 per second (or 44.1 kHz.) The most common PCM data structure for DVDs is "24/96" or 24-bit words at the rate of 96,000 per second (or 96 kHz).

Phase
The relative timing of two or more signals. Signals reproduced in time with each other are said to be "in phase" while signals reproduced out of synchronization with each other are said to be out of phase to one degree or another.

Properly "phased" loudspeakers, where each loudspeaker system in an audio/video system emits a positive pulse when the amplifier delivers one, are necessary for proper spatial reproduction of music or movie soundtracks. In addition, "Phase" is central to understanding how a "matrix" works.

Pink Noise
A type of equalized random noise which contains equal amounts of energy in each of the ten audible octaves. Extensively used in audio testing, it resembles the energy distribution found in music.

Planar driver
A generic term for a comparatively large, flat driver that most often is as large as the height and width of the speaker itself. Planar drivers, when properly excited by an amplifier's signal, move air equally well from all points – front and rear – on their surface. (Dynamic drivers, in contrast, are small, radiate in only one direction, and are the only active elements on a speaker's baffle.)

Polypropylene/Polystyrene Capacitors
These very high quality components are usually employed in the signal path where large value capacitors are not required. Their advantage is that they have a very small "Miller Effect" or memory, compared to electrolytic capacitors. This means that each subsequent signal passing through them has greatly reduced coloration, resulting in better fidelity.

Power Amplifier

An electronic circuit or component that increases the level of signal from a source or preamplifier until it is able to drive a loudspeaker.

Preamplifier

An audio/video system's switching center, a preamplifier allows the user to choose a particular source component (CD or tuner, for example), adjust tonal quality and system volume.

Precedence Effect

A phenomenon of human hearing by which we judge the origin of a sound coming from more than one direction by where we first hear that sound coming from. For example, if a car on your left honks its horn, the sound reaches your left ear before it reaches your right ear, and using this information, your brain concludes that the car is on your left. Also known as the Haas Effect.

Radiation Pattern

The area over which a loudspeaker radiates most of its energy and how evenly this energy is distributed within that area. Speakers may be omnidirectional and radiate freely in all directions or directional with specifically designed radiation patters. Directional speakers can be broken down further into three main groups: direct radiators (energy distributed primarily to the front of the speaker), bipolar and dipolar (equal energy radiation to the front and rear of the enclosure but phased differently.

Receiver

An audio or audio/video component that combines a tuner, preamplifier and power amplifier in a single chassis.

Sensitivity

Has many meanings in audio/video:

1) The measure of how well a tuner circuit will pick up a weak signal. This is usually measured in microvolts or dBf (decibels per femtowatt). Lower numbers are usually better.

2) The measure of how loud a speaker will play with a standard input voltage applied at the terminals. Sensitivity is usually expressed as "XX dB SPL for 1 watt input (or 2.83 volts, the equivalent at 8 ohms) measured at 1 meter." Sensitivity is no measure of a speaker's sound quality, only a reflection of how loud it will play with a given input. (See also: Efficiency.)

Single-beam laser transport

An optical laser assembly that uses a single beam to read the "bumps and pits" off a compact disc (CD). (See also: Three-beam laser transport.)

Slit Foil Capacitors

These specially designed power supply capacitors are manufactured for Rotel in Great Britain. Their unique properties feature a low dielectric loss with extremely fast time constants (charge/discharge time) in relation to their storage capacity. In operation, these capacitors emulate the design achievements of many esoteric products which use a very large number of smaller capacitors to achieve much the same effect, albeit at substantially higher cost.

SMPTE

Society of Motion Picture & Television Engineers

Star Earth Grounding

A technique used by designers to ensure no degradation of sound or acquired "ground loops" by poor "earthing" arrangements. Star earth grounding or "Y" connection, routes all ground wires to a single point in the design with the resulting wire arrangement looking somewhat like a star pattern.

Subwoofer

A separate speaker designed specifically for the accurate reproduction of low frequency information

Symmetrical Circuit Layout

A method of circuit configuration that lays out both left and right halves of the design in a "mirror image" of the other. The result is matched electrical performance from both channels for superb stereo resolution.

Thermal noise

The very faint background noise added to the signal as electrons pass through semiconductors and cause heat.

Three-beam laser transport

A three beam optical pick-up passes light from the laser source through a diffraction grating resulting in 3 separate beams. The center beam is used to read data and feeds information to keep the beam focused. The two other beams are used for tracking. (See also: Single-beam laser transport.)

THX

This is set of certification standards established by Lucasfilm Ltd. that were set-up in order to ensure that what you hear in your home or theater comes as close as possible to what the soundtrack engineer originally intended you to hear. The technology works hand in hand with Dolby Pro Logic or Dolby Digital circuitry. In short, equipment that bears this trademark must meet certain minimum standards of performance. Decoders must further incorporate special circuitry that re-equalizes the high frequencies to compensate for the differences between theater and home playback. Further, they must include Timbre Matching, an EQ circuit that smoothes the transitions between front and surround speakers.

Toroidal Transformer

A donut shaped transformer that provides higher efficiency and lower stray magnetic fields for its size, relative to an equivalent sized, standard iron-core transformer.

Transducer

Any device that converts one form of energy into another. A solar panel, for example, is a transducer because it converts sunlight into electricity. Almost all loudspeakers are actually "double transducers" in that they first convert electrical energy (from an amplifier) into mechanical energy (a piston moving back and forth, for example) and finally convert that mechanical energy into acoustical energy (sound waves) that we can hear. Both types of energy conversion (electrical → mechanical and mechanical → acoustical) are very important design factors in any loudspeaker.

Tuner

An audio or video component designed to receive radio or television broadcasts.

WG-4

"Working Group 4" is the DVD Forum committee responsible for developing the audio standards for DVD.

NOTES