# MPEG-2:

## The basics of how it works

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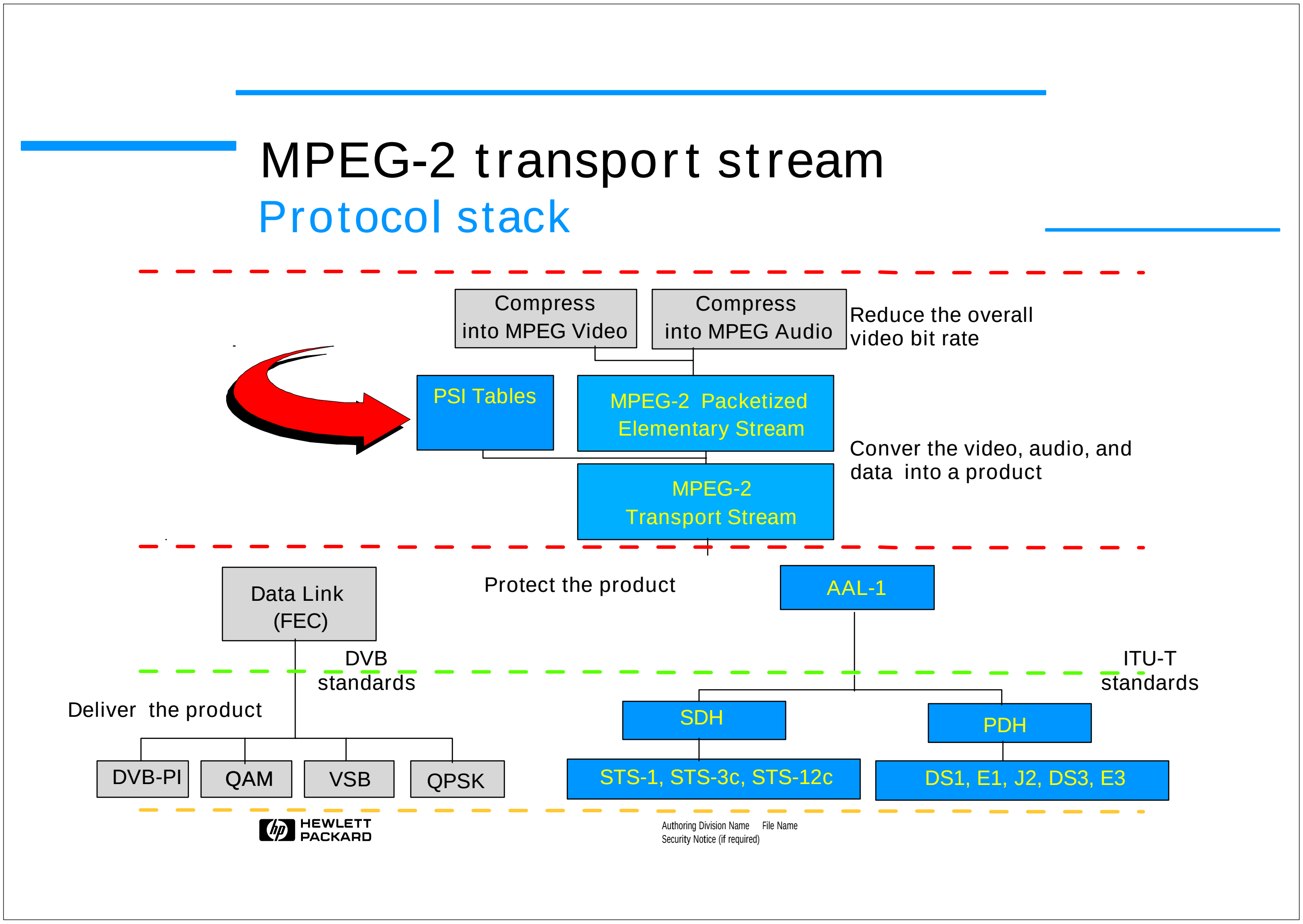
This slide & speaker notes based tutorial, covers the very basics of how MPEG-2 works. That is how a high bit rate digital video signal is taken from a studio camera. Compressed down to a low enough bit rate, so  that it can make economic use of available transmission bandwidth. And converted into a form that a consumer set-top box can decode.

The tutorial covers the creation of the elementary stream, the packetised elementary stream, and the transport stream multiplex.
It discusses the use of Programme Specific Information, and also the extensions created by the DVB, known as the DVB Service Information.

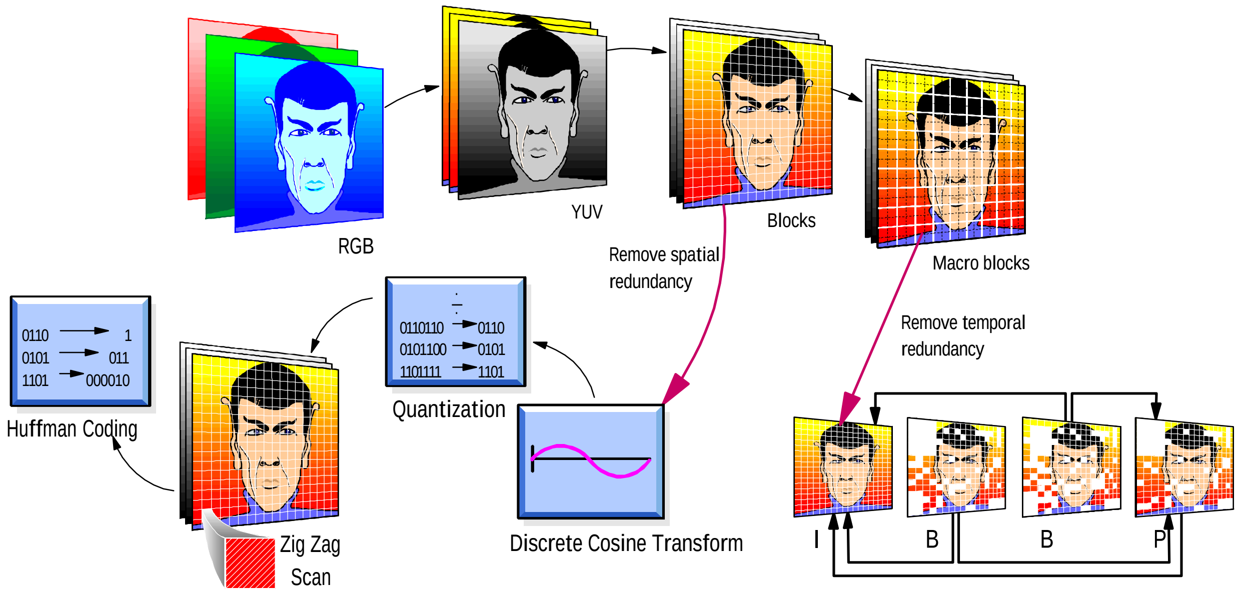
It also tries to give a feel for the fragility of the MPEG-2 transport stream. This is of major importance to service providers, wishing o guarantee quality of service.




- This slide shows the protocol levels that are used to transport the digital video signal, from the studio to the consumer.
- We are interested in the section between red dashed lines. Transport stream protection is covered in another tutorial.
- The protected transport stream is then fed into whichever physical channel is used to deliver the video signal to the consumer.
- Note this architecture is point to multi-point, constant bit rate, broadcast orientated: which is why only AAL-1 is shown, not AAL-5.

# MPEG compression of video:

## Spatial & temporal redundancy

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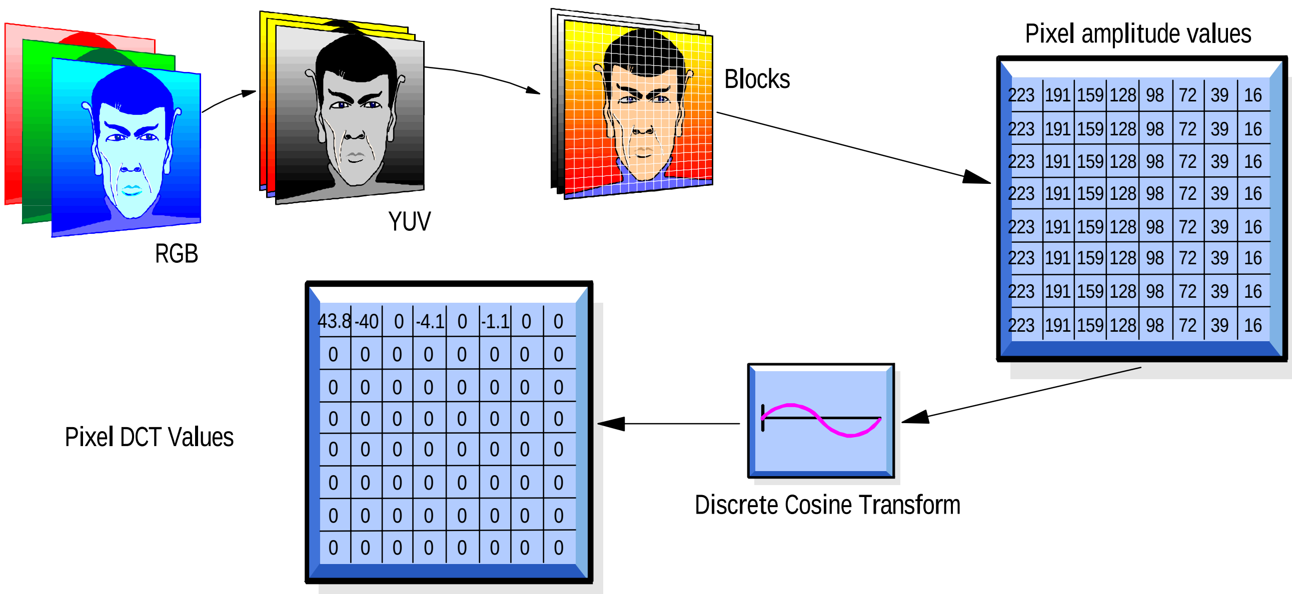
- Video compression relies on the eye's inability to resolve High Frequency color changes, and the fact that theres a lot of redundancy within each frame and between frames.
- The Discrete Cosine Transform is used, along with quantization and Huffmann coding; to predict a pixel value from all adjacent pixel values, and minimize the overall bit rate.
- This generates the Intra-frames (I-frames).
- Prediction & motion compensation, predicts the value of pixels in a frame, from the information in adjacent frames.
- Audio compression makes use of the fact that, high power tones tend to blot out lower power adjacent tones. So if you can't hear it, don't transmit it.

# Spatial redundancy:

## Pixel coding using the DCT




Pixel amplitude values

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |
| 223 | 191 | 159 | 128 | 98 | 72 | 39 | 16 |

Pixel DCT Values

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 43.8 | -40 | 0 | -4.1 | 0 | -1.1 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |

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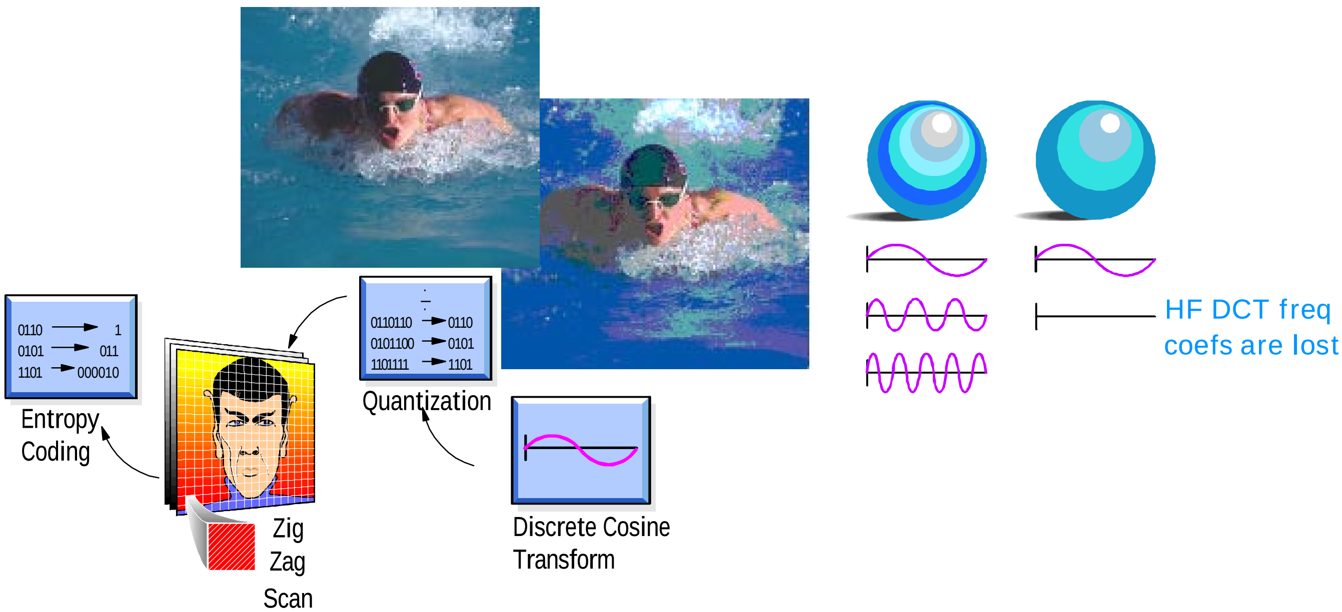
- The first stage is to create an I-frame, subsequent frames in a group of frames will be predicted from this frame.
- As the eye is insensitive to HF color changes, we convert the R,G,B signal into a luminance (how bright the picture is) and two color difference signals. We can remove more U,V information than Y.
- Each pixels DCT is calculated from all other pixel values, so taking 8x8 blocks reduces the processing time.
- The top left pixel in a block is taken as the dc datum for the block.
- DCT's to the right of the datum are increasingly higher horizontal spatial freqs. DCT's below are higher vertical spatial frequencies.
- Using an Inverse DCT we could reconstruct each pixel's value in the 8x8 block. The DCT is a lossless and reversible process.
- Its the next stage which introduces compression.
- Note that the smaller the difference between one pixel and its adjacent pixels, the smaller its DCT value.
- In the example shown, a greyscales 8x8 pixel values are reduced to one row of DCT's. With all other values going to zero

# Spatial redundancy:

## Quantization & Entropy coding

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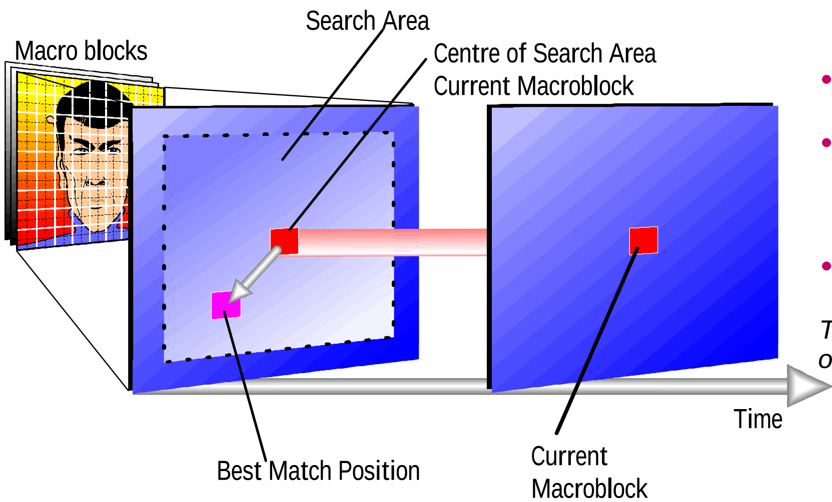
- The higher the DCT frequency, the higher the Quant Matrix value its divided by. This makes many coeficientss go to zero.
- The fixed value scale factor reduces even more of the DCT's to zero.
- The next stage is to increase the number of zero's in the run of bits into the entropy coder. This is done by zig-zag scanning the 8x8 pixel block DCT values and helps the entropy coder do its job.
- Entropy coding essentially sizes coeficients by how often they occur.
- The more a coeficient occurs, the smaller a binary value its given.
- Since in any frame your going to get a large number of identical 8x8 blocks, your reducing the overall binary data rate.
- To summarize then, quantization makes many higher frequency DCT values go to zero. Entropy coding removes duplication of DCT's, assigning each DCT position with a pointer to its value.
- This all has a cost. Thats shown  in the pictures above: the upper picture is unquantized, the lower one quantized.

# Temporal redundancy:

## Inter-frame prediction & motion estimation




- If adjacent frame macroblocks same, don't retransmit block.
- Search to see if same block exists somewhere else: if it does, just transmit its coordinates (motion vectors).
- Only Intra-code macroblocks which are completely new.

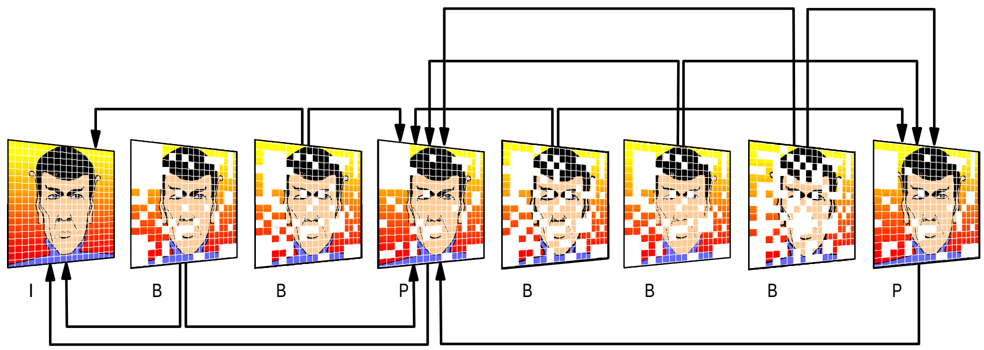
*This process really drops the overall bit rate*



- This is were the real bit rate reduction kicks in. As we'll cover in the next slide, there are three different frame types.
- By just doing spatial redundancy on a frame you create an I frame. This has all the information necessary to decode the picture.
- The next stage is to look at the next frame to this and see how similar it is. You can do three things to minimize this frames bit rate.
- Firstly, look to see if the macroblock in the same position in the next frame hasn't change. If it hasn't, don't do any coding, Just transmit that its the same.
- The next stage is to search around in the I-frame and see if this macro-block exists, but its in a different place. If so transmit motion vectors for its old location.
- Only if its completely new, do you go for the complete intra-coding process.
- This really reduces the overall bit rate from frame to frame.
- But note if you kept predicting each frame from the last, it would only take a little error, and the whole process would fast start to unravel.
- Thats why there are three different frame types, and a specific frame transmit process.

# Putting it all together

## I, P, B Frames & avoiding "MPEG heat death"




- I-frames: contain full picture information
- P-frames: predicted from past I or P frames
- B-frames: use past and future I or P frames
- Transmit I frames every 12 frames or so.

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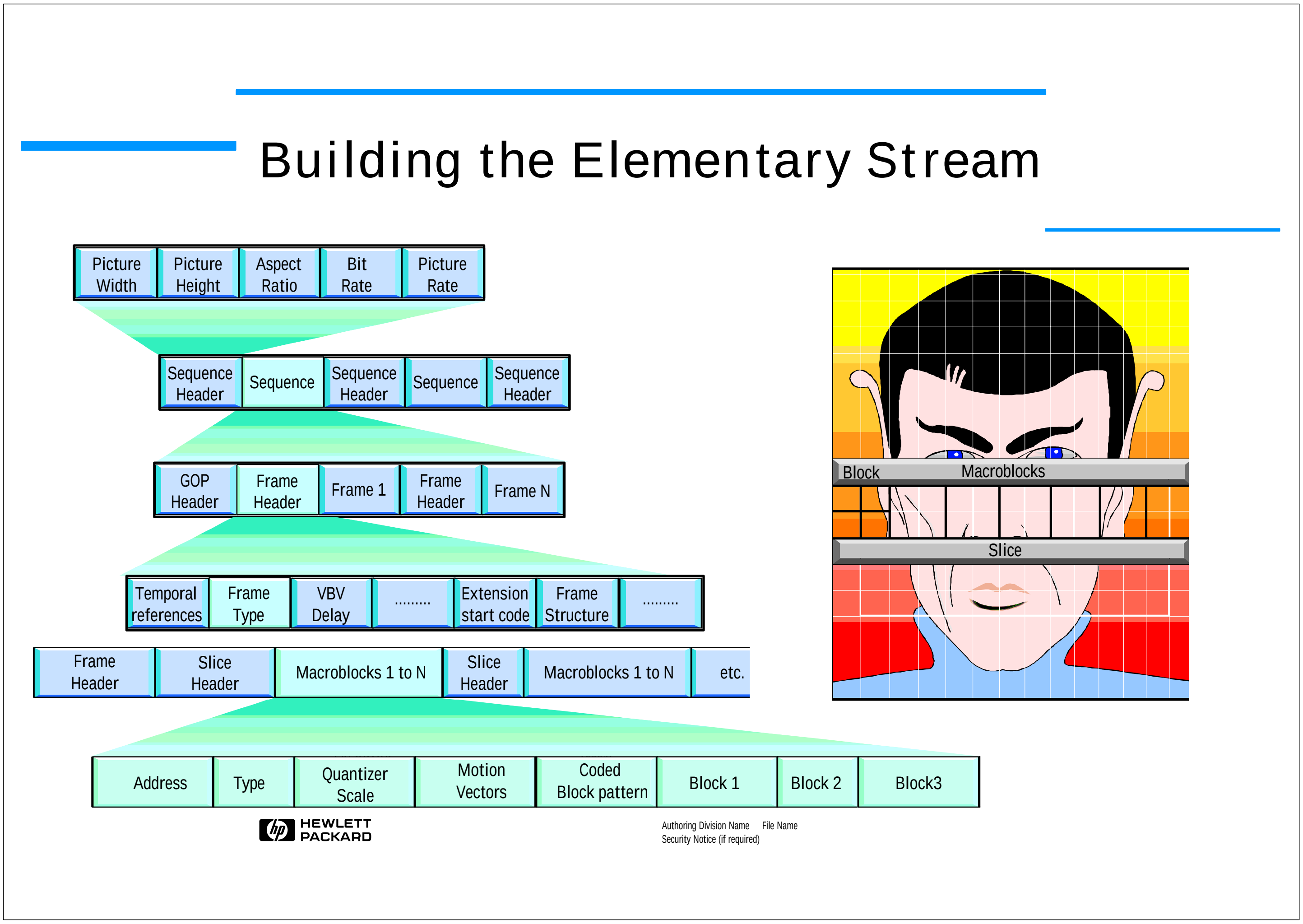
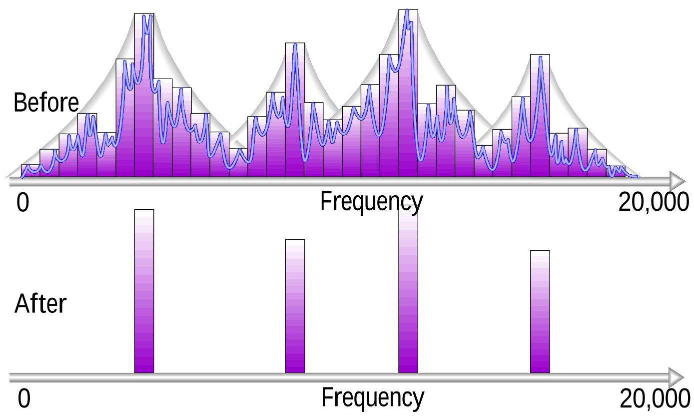
- The Intra Frames contain full picture information. These are your lifeline, if errors occur, or the decoder loses a frame. Without periodic transmission of these the whole process falls apart. But the I-frames are the least compressed.
- Predicted (P) Frames are predicted from past I, or P frames,
- Bi-directional predicted frames offer the greatest compression and use past and future I & P frames for motion compensation. But they are the most sensitive to errors.
- The encoder will cycle through each frame and decide whether to do I, P, or B coding. The order will depend on the application. But roughly every twelve frames, an I-frame is created.
- If the encoder didn't do this, any small errors would build up and the MPEG compressor would rapidly descend into an electronic form of Entropic "heat death".
- The process detailed in the last few slides does the real work. But a decoder needs additional information to reconstruct the frames.




- This slide shows how the actual blocks, slices, frames etc. are all put together to form the elementary stream.
- Along with the actual picture data, header information is required to reconstruct the I, B, P frames. This header structure is shown.
- Each slice will contain a header detailing its contents & location.
- Each frame will have a header, and each group of I, B, P frames, known as a Group Of Pictures (GOP) will have a header.
- The next stage is to take this ES and convert it into something that can be transmitted and decoded at the other end.
- At this stage, the elementary stream is a continual stream of encoded video frames. Though all the data required to reconstuct frames exists here. No timing information or systems data is contained
- Thats the job of the MPEG-2 multiplexer
- First a few words on what we do with the audio signal associated with the video

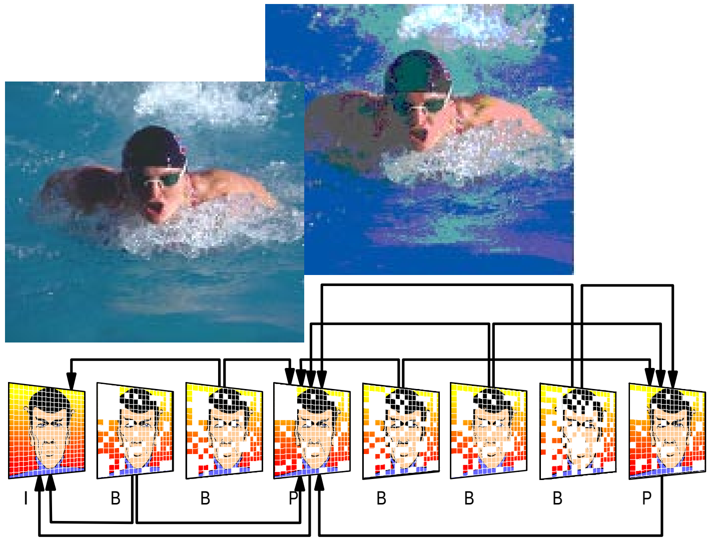
# Sub-Band Masking

- System relies on masking of tones either side of hi-power tone.
- If you can't hear it, don't code it.
- Sub-band masking takes 3 MBit/s & reduces it to about 200 Kbit/s.
- Two types of audio coding: exists MPEG-2 & Dolby AC3
- US use AC3, DVB uses MPEG-2.



# MPEG compression: summary

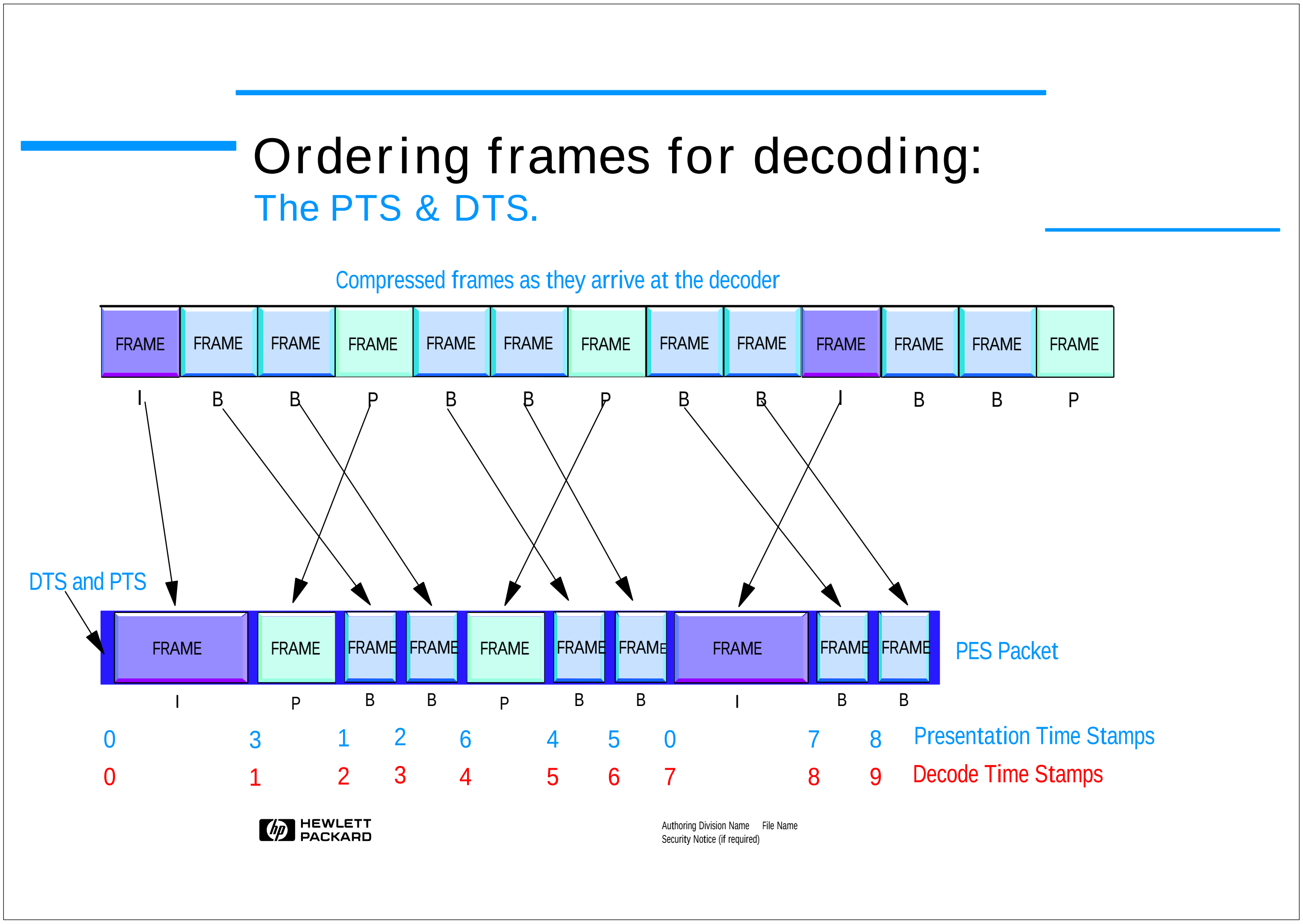
## Reduced bit rate, increased error susceptibility




*But how do we know what channel we've compressed?*

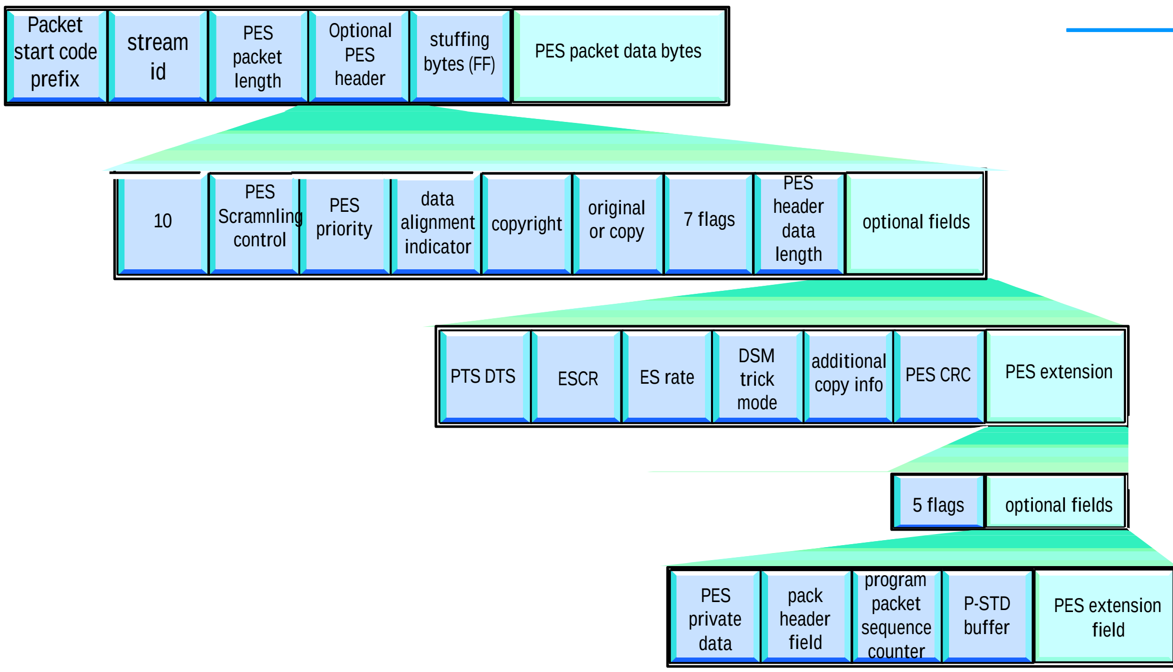


- This process gets rid of a huge amount of redundant information.
- But that means we've lost information we could use if errors occur.
- The compression process may make digital tv economically viable, but its already caused a problem for the decoder, if errors are injected into the data: Quantization & motion estimation ensure, these errors quickly propagate.
- Unfortunately it gets worse.
- There's no point doing all this compression alone. Any decoder has to know what information you've compressed, what channel it is, is there any special data contained. And a lot of other housekeeping information.
- Thats why you need a systems layer. Thats what we'll describe next.
- The QoS problem is already getting big due to compression. But at least at this level you only lose some frames. At the systems level, you will lose many frames if errors occur.




- In order for a decoder to reconstruct a B-frame from the preceding I and following P frames, both these must arrive first.
- So the order of frame transmission must be different to the order they appear on the tv screen. As shown in the slide above.
- But for this to work, the decoder must also know at what time it should show the frames. That its their order in time.
- The Decode Time Stamp tells the decoder when to decode the frame.
- The Presentation Time Stamp tells the decoder when to display the frame.
- In addition to knowing at what time decode and presentation should occur, a clock must be embedded, to allow a time reference to be created.
- The PTS and DTS are added to the Packetised Elementary Stream, whilst the clock, known as the Programme Clock Reference (PCR) is contained in the Transport stream.
- The PCR will be discussed shortly.

# The Packetised Elementary Stream

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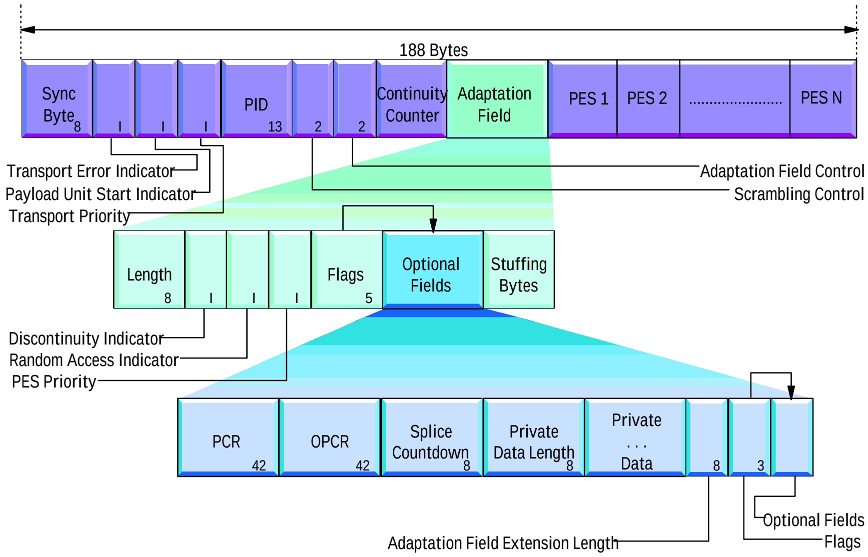
- This slide shows the PES structure.
- The PES header contains information about the content of the PES packet data bytes. Allowing a decoder to make sense of the packets.
- The PES packets can be of variable length, typically upto 64 kbytes, but they can be longer.
- One of the most important parts of the structure, are the PTS and DTS, these allow the decoder to reconstruct the video stream from the I, B, P frames sent by the encoder.
- The PTS & DTS process was discussed in the previous slide.
- The important thing to note is that if information carried in the header is corrupted, the entire PES packet will be lost.
- As seen, the higher up the MPEG protocol stack we go, the greater the potential damage that errors will cause.
- The next stage is to multiplex many PES together to create a stream of many tv programme events.
- Since at the moment, this is simply a raw, but decodable single elementary stream: without associated audio, or data. And no infomation which tells the consumer what it contains.

# MPEG-2 Transport Stream

## multiplexing many programs

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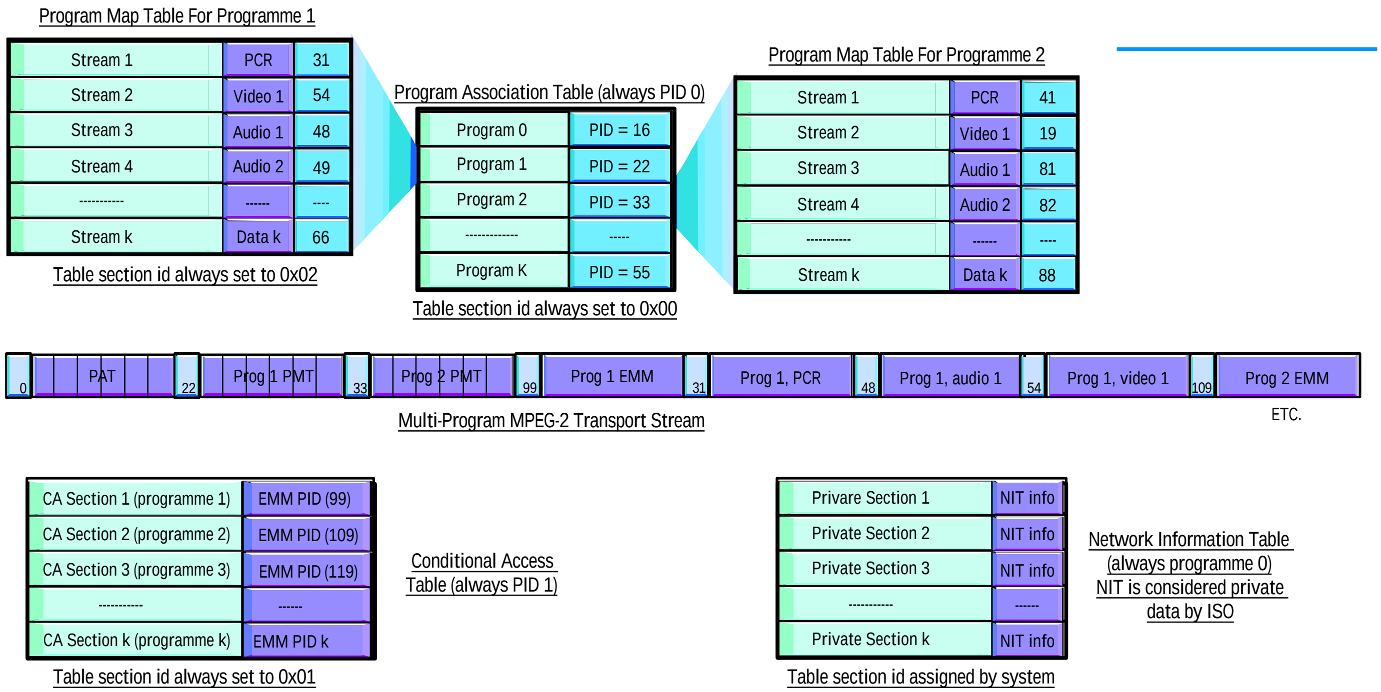
- This is a blowup of the Transport Stream packet structure
- The TS takes the variable length PES, and chops it up into fixed length packets: this is required for most transmission systems.
- Note this information is on top of the syntax information contained in the PES header. Perhaps you begin to get a feel for the error potential here. The TS allows the multiplexing of many PES.
- The key features are as follows:
- Sync Byte, sets the start of a TS packet and allows transmission synchronization.
- Transport Error indicator: indicates the packet is errored (block error testing)
- PID: Packet IDentifier, is the channel identifier It contains all the navigation information required to find, identify and reconstruct programmes. PID values are contained in the PSI tables.
- PCR, the programme clock reference: provides 27 MHz clock recovery information.
- How does the decoder know which packets containf the PES, which makes up the tv event to decode?

# PSI Table structure

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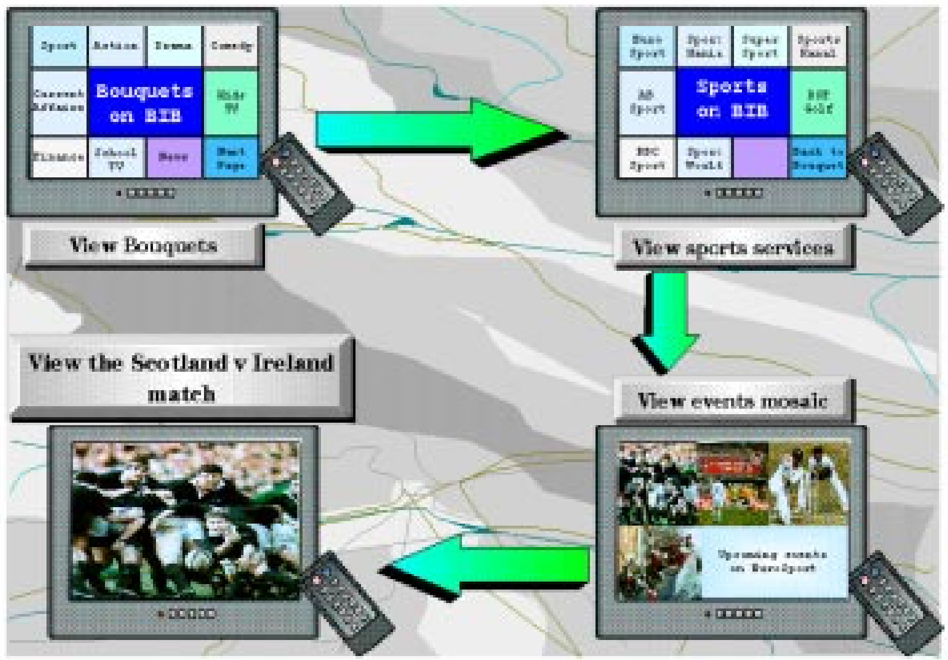
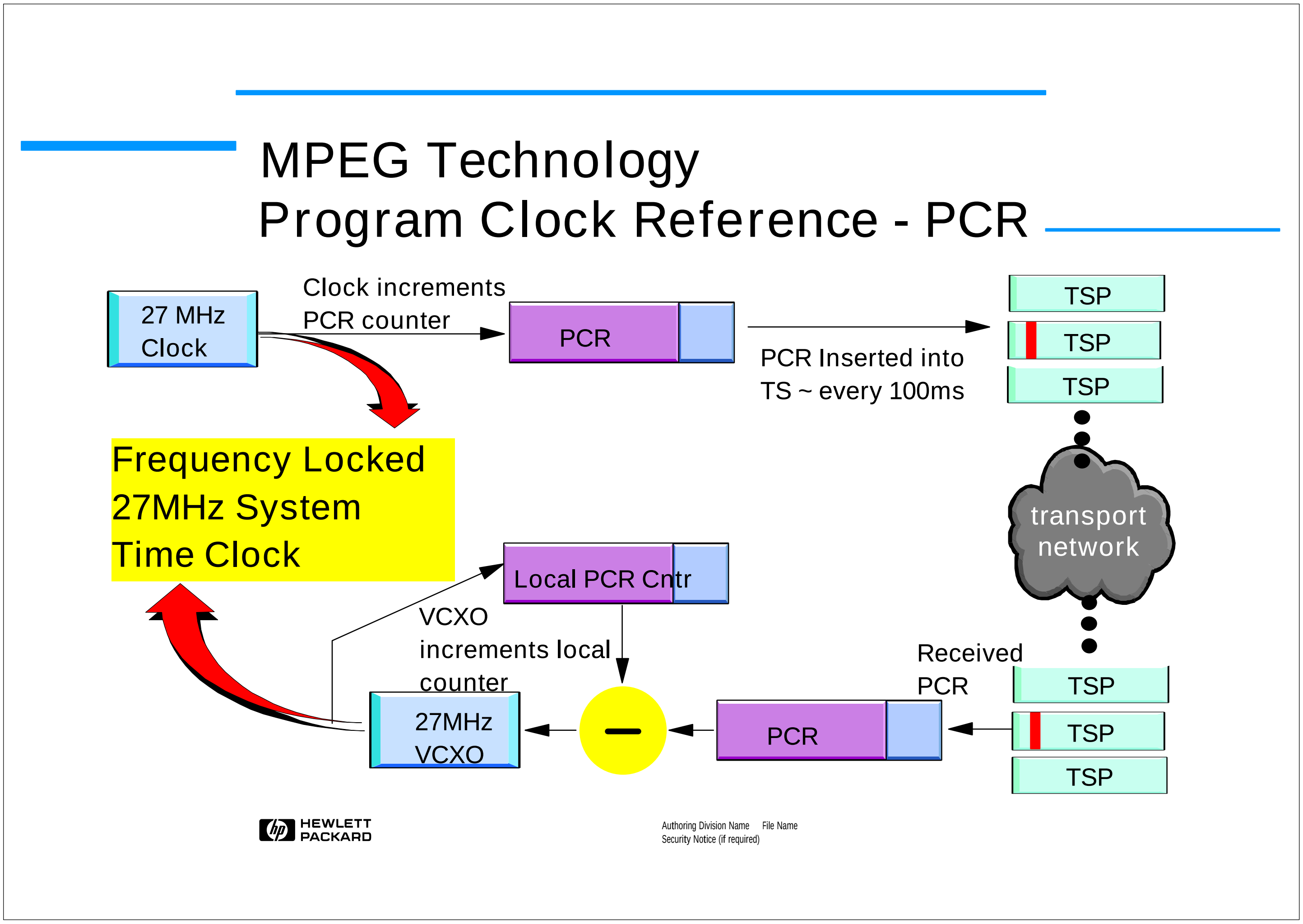
- To reconstruct the PES, the PSI uses a series of identifiers known as the Programme Identifiers, or PID's.
- Once the programme to be decoded is known, the decoder searches for PID=0 - the Programme Association Table PID.
- The PAT contains the PID's of all the PMT's for the programmes contained. We assume programme one has been chosen for decode.
- The PMT for programme 1 is identified via its PID (22), extracted from the transport stream packets containing it, and decoded.
- Prog 1's PMT contains all the PID's for Prog1's video, audio and data packets. These must be put together to reconstruct the PES.
- Prog 1's timing info, required for decode, is contained in a transport packet, identified by the PCR PID (31). Each prog has a PCR.
- PID zero is always used to identify the CAT. This is needed to find out whether the consumer is allowed to decode and view prog 1.
- The CAT contains all the PID's identifying the EMM's for all progs.
- The NIT, contains information about the user-selected service. Such as channel freq's transponder numbers etc. The NIT is always associated with prog 0's PID. See DVB-SI for more about the NIT.

# DVB-SI: program guides



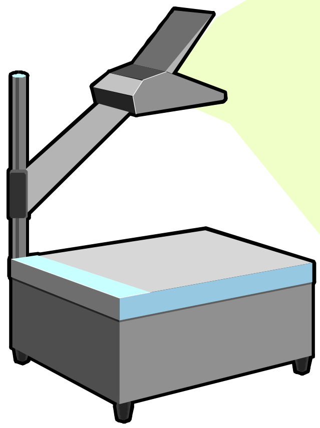



- WIthin the PSI structure, is a table known as a private table. This was created by MPEG, so that service providers could create their own extentions to the MPEG-2 PSI.
- This has been used by the DVB and the ATSC, to create what are called Service Information tables. We discuss the DVB-SI only.
- These are used to carry information about what tv events are contained in the transport stream: via the Event Information Table.
- What services are being carried, via the Service Desciption Table.
- What groups of services, with common themes exist: via the Bouquet Association Table.
- And what are the physical parameters of the network carrying a transport stream: via the Network Information Table.
- The tables are highly complex constructs. And like the PSI, are protected by a CRC-32.
- The SI are used to create Electronic Programme Guides (EPG's). These help the consumer find the programme they want to watch.
- The example shows the process by which a user goes from the display of bouquet's all the way to viewing a Rugby match.




- PCR is a clock recovery mechanism for MPEG programs. When a program is encoded, a 27MHz System Time Clock (STC) drives the encoding process. When the program is decoded (or remultiplexed), the decoding process must be driven by a clock which is locked to the encoder's STC. The decoder uses the PCR to regenerate a local 27MHz clock.
- When a program is inserted into the Transport Stream a 27MHz timestamp is inserted - the PCR. At the decoder end, it uses a Voltage Controlled Oscillator (VCXO) to generate a 27MHz clock. When a PCR is received, it is compared to a local counter which is driven by the VCXO, and the difference is used to correct the frequency of the VCXO to ensure that the 27MHz clock is locked to the PCR.
- The PCR field is a 42 bit field in the adaptation field of the Transport Stream. The PCR field consists of a 9 bit part that increments at a 27MHz rate and a 33 bit part that increments at a 90kHz rate (when the 27MHz part rolls over).

## Summary: *"the MPEG-2 transport stream is efficient, but fragile"*



- Corrupt ES headers lose frames
- Corrupt PES header, lose whole ES
- TS carries many PES & own hdr (4byte)
- Hdr contains info on navigation thro PES (PID &PSI). Corrupt this, lose many PES
- Corrupt SI & EPG: lose pay-per view
- PCR provides timing clock. PCR errors means you lose timing & PTS/DTS.

*"theres no margin for error, the TS needs protection"*

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- The elementary stream is taken and formed into a Packetised Elementary Stream. This has a header which allows ES decode.
- The TS carries many PES, along with the PSI, SI and a header.
- If the TS header is corrupted, then many PES will be lost.
- If the PSI, or SI are corrupted, then it may be impossible for the consumer to find out what is contained in the transport stream.
- To conclude, though MPEG-2 allows the economic transmission of many different tv events, along with associated service information, it does so by having very little margin for error.
- Due to this, the protection of the transport stream, and the ability to verify its integrity is vital.
- The problem with tv broadcast is any errors are glaringly obvious, and directly affect the consumer. This means the potential for service provider conflict is high.

# HYPOTHALAMUS AND AUTONOMIC NERVOUS SYSTEM

## A. Hypothalamus = Homeostasis

The main function of the hypothalamus is **homeostasis**, or maintaining the body's status quo. Factors such as blood pressure, body temperature, fluid and electrolyte balance, and body weight are held to a precise value called the set-point. Although this set-point can migrate over time, from day to day it is remarkably fixed.

To achieve this task, the hypothalamus must receive inputs about the state of the body, and must be able to initiate compensatory changes if anything drifts out of whack. The inputs include:

- **nucleus of the solitary tract -** this nucleus collects all of the visceral sensory information from the vagus and relays it to the hypothalamus and other targets. Information includes blood pressure and gut distension.

- **reticular formation -** this catchall nucleus in the brainstem receives a variety of inputs from the spinal cord. Among them is information about skin temperature, which is relayed to the hypothalamus.

- **retina -** some fibers from the optic nerve go directly to a small nucleus within the hypothalamus called the **suprachiasmatic nucleus**. This nucleus regulates circadian rhythms, and couples the rhythms to the light/dark cycles.

- **circumventricular organs -** these nuclei are located along the ventricles, and are unique in the brain in that they lack a blood-brain barrier. This allows them to monitor substances in the blood that would normally be shielded from neural tissue. Examples are the **OVLT**, which is sensitive to changes in osmolarity, and the **area postrema**, which is sensitive to toxins in the blood and can induce vomiting. Both of these project to the hypothalamus.

- **limbic and olfactory systems -** structures such as the amygdala, the hippocampus, and the olfactory cortex project to the hypothalamus, and probably help to regulate behaviors such as eating and reproduction.

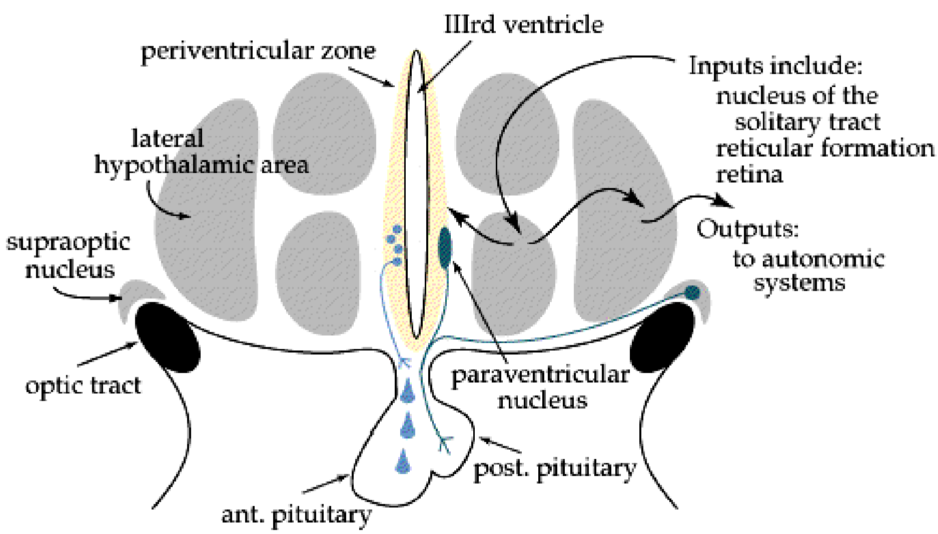
The hypothalamus also has some intrinsic receptors, including **thermoreceptors** and **osmoreceptors** to monitor temperature and ionic balance, respectively.

Once the hypothalamus is aware of a problem, how does it fix it? Essentially, there are two main outputs:

- **neural signals to the autonomic system -** the (lateral) hypothalamus projects to the (lateral) medulla, where the cells that drive the autonomic systems are located. These include the parasympathetic vagal nuclei and a group of cells that descend to the sympathetic system in the spinal cord. With access to these systems, the hypothalamus can control heart rate, vasoconstriction, digestion, sweating, etc.

- **endocrine signals to/through the pituitary -** recall that an endocrine signal is a chemical signal sent via the bloodstream. Large hypothalamic cells around the third ventricle send their axons directly to the **posterior pituitary**, where the axon terminals release **oxytocin** and **vasopressin** into the bloodstream. Smaller cells in the same area send their axons only as far as the base of the pituitary, where they empty **releasing factors** into the capillary system of the **anterior pituitary**. These releasing factors induce the anterior pituitary to secrete any one of at least six hormones, including **ACTH** and **thyroid-stimulating hormone (TSH)**.

Ultimately the hypothalamus can control every endocrine gland in the body, and alter blood pressure (through vasopressin and vasoconstriction), body temperature, metabolism (through TSH), and adrenaline levels (through ACTH).



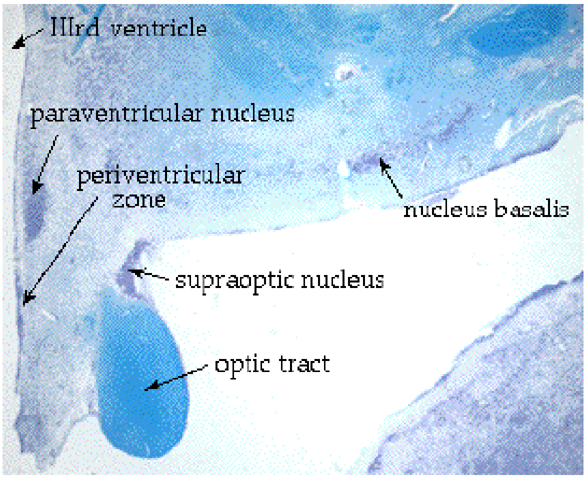

**In the news lately:** The hypothalamus controls body weight and appetite, but it is not entirely clear how. Sensory inputs, including taste, smell, and gut distension, all tell the hypothalamus if we are hungry, full, or smelling a steak. Yet it is mysterious how we are able to vary our eating habits day to day and yet maintain about the same weight (sometimes despite all efforts to the contrary!) . The "set-point" theory is an old one in diet science, but until recently the mechanics of maintaining that set point were unknown. It appears that there is an endocrine component to the appetite system. Recent studies in mice have shown that the fat cells of normal overfed mice will release a protein called

**leptin** (or **OB**, after the gene name), which reduces appetite and perks up metabolism. Leptin is presumably acting on the hypothalamus. Underfed mice, on the other hand, produce little or no leptin, and they experience an increase in appetite and a decrease in metabolism. In both of these mice, the result is a return to normal weight. But what would happen if a mouse (or human) had a defective OB gene? Weight gain would never trigger fat cells to release leptin, the hypothalamus would never slow the appetite or increase metabolism, and the mouse would slowly but surely become obese (how the gene got its name). Sure enough, shortly after these experiments hit the news, the human OB gene was discovered and a few obese patients were found to have the mutation. Many more obese patients had normal OB genes, however, indicating that there is much more to the story yet to be discovered.

**B. The anatomy of the hypothalamus:**

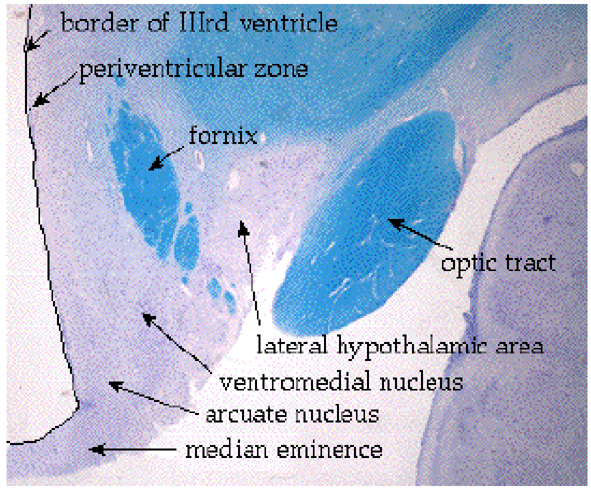
The hypothalamus, as you would expect from the name, is located below the thalamus on either side of the third ventricle. These sections have been cut coronally, and show only one side of the hypothalamus.

In this anterior section through hypothalamus, you can see the large neurons of the paraven-tricular nucleus, which send axons to the posterior pituitary. The cells in the periventricular zone send axons to the median eminence, from which releasing factors are carried to the anterior pituitary.



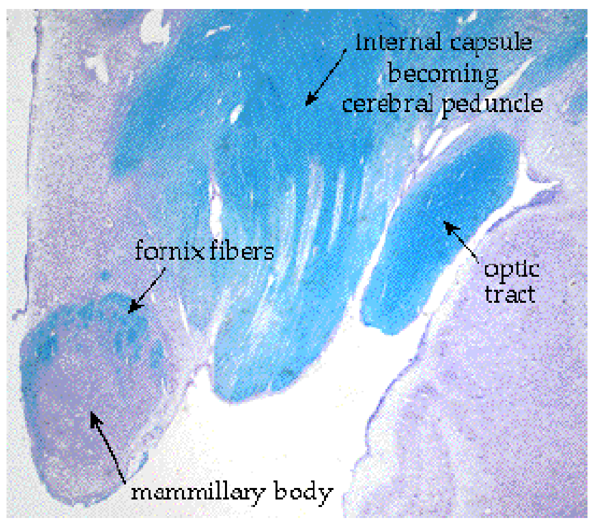

The nucleus basalis is a cholinergic nucleus involved in sleep and wakefulness.

This section is posterior to the first. The hypothalamic nuclei are hard to distinguish, but the arrows point out approximate locations. The pituitary stalk would normally be continuous with the median eminence, but it is a fragile structure usually lost in dissection.




Note the fornix descending through the hypothalamus. The fornix originates in the hippo-campus and ends in the mammillary bodies.

In this posterior section you can see the fornix joining the mammillary body. This is also a nice section to demonstrate the way that the internal capsule fibers flow into the cerebral peduncle.




**C. The autonomic nervous system:**

The autonomic nervous system is an entire little brain unto itself; its name comes from "autonomous", and it runs bodily functions without our awareness or control. It is divided into two systems which, where they act together, often oppose each other: the **sympathetic** and **parasympathetic** systems. The sympathetic system evokes responses characteristic of the "fight-or-flight" response: pupils dilate, muscle vasculature dilates, the heart rate increases, and the digestive system is put on hold. The parasympathetic system has many specific functions, including slowing the heart, constricting the pupils, stimulating the gut and salivary glands, and other responses that are not a priority when being "chased by a tiger". The state of the body at any given time represents a balance between these two systems.

The best way to learn the functions and structures of each system is by comparison. The following table lists some attributes of each:

| **The Parasympathetic System** | **The Sympathetic System** |
|---|---|
| Origins:<br><br>Parasympathetic cells are located in different nuclei throughout the **brainstem**, as well as a few in the **sacral spinal cord**. Their axons travel to the target organ, synapse in **ganglia** in or near the organ wall, and finally innervate the organ as "post-ganglionics". Examples of these ganglia include the ciliary, otic, and pterygopalatine ganglia in the head, and diffuse networks of cells in the walls of the heart, gut, and bladder. | The cells of the **intermediolateral column** in the **thoracic spinal cord** are the source of all the sympathetics. They also travel to ganglia before reaching the target organ, but the sympathetic ganglia are often far from the target. Some notable ganglia: |
| Nuclei of origin:<br><br>**Edinger- Westphal nucleus -** Axons from this nucleus travel with cranial nerve III and have 2 functions:<br>- pupil constriction<br>- lens accommodation | **Superior cervical ganglion -** supplies sympathetics to the head, including those that:<br>- dilate the pupils<br>- stimulate sweat glands<br>- lift the eyelids |
| **Salivatory nuclei -** These nuclei in the medulla send axons to the salivary glands via the VIIth and IXth nerves. | **Celiac and mesenteric ganglia -** These ganglia distribute sympathetics to the gut. Functions include:<br>- vasoconstriction<br>- inhibition of secretions |
| **Dorsal nucleus of the vagus -** | |

This nucleus gives rise to the secretomotor fibers of the vagus nerve (X). Its functions include:
- stimulate gastric secretion
- stimulate gut motility
- stimulate respiratory secretions

**Nucleus ambiguus** (and surrounding cells) - Axons from these cells project via the vagus to the heart, lungs, and pharynx. Functions include:
- decrease heart rate
- bronchial constriction

**Chain ganglia** running along the spinal cord distribute sympathetics to the thorax and periphery to:
- increase heart rate
- dilate bronchi
- selectively vasoconstrict
- vasodilate in active muscles

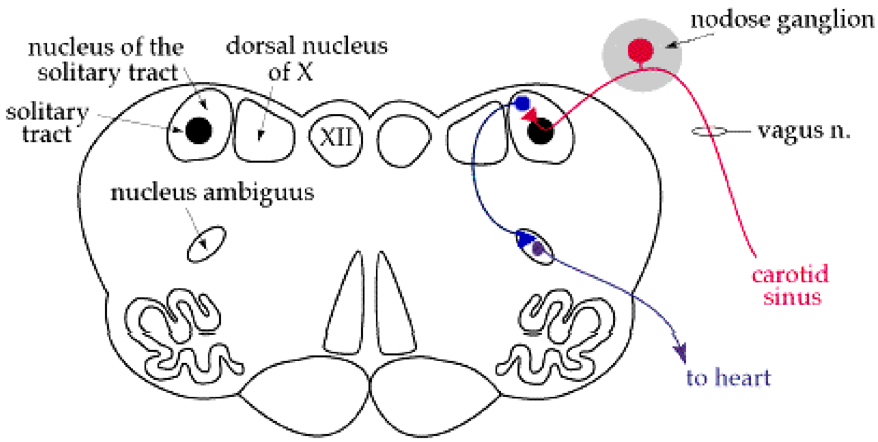
The autonomic system also receives **afferents** that carry information about the internal organs. They return to separate locations:

Parasympathetic afferents
Nearly all of the afferents return via the vagus to a single nucleus, the **nucleus of the solitary tract**. Like all sensory afferents, the actual cell bodies of the neurons sit just outside the CNS in a ganglion (the **nodose ganglion**). The central processes of the neurons enter the medulla in the **solitary tract** and travel a bit before synapsing in the surrounding nucleus of the solitary tract. The solitary tract is somewhat analogous to Lissauer's tract in the spinal cord.
The nucleus receives information about blood

Sympathetic afferents
Afferents reenter the dorsal horn of the spinal cord along side of the sensory afferents from the skin. The sympathetic afferents mainly carry information about visceral pain. Since this information converges with pain from the body surface, the pain is often perceived as originating at the body surface instead of deep in the viscera. This phenomenon is called **referred pain**, and follows predictable patterns. For example, afferents from the heart enter the spinal cord at the same level as those from the shoulder region. This is why pain in the heart (a heart

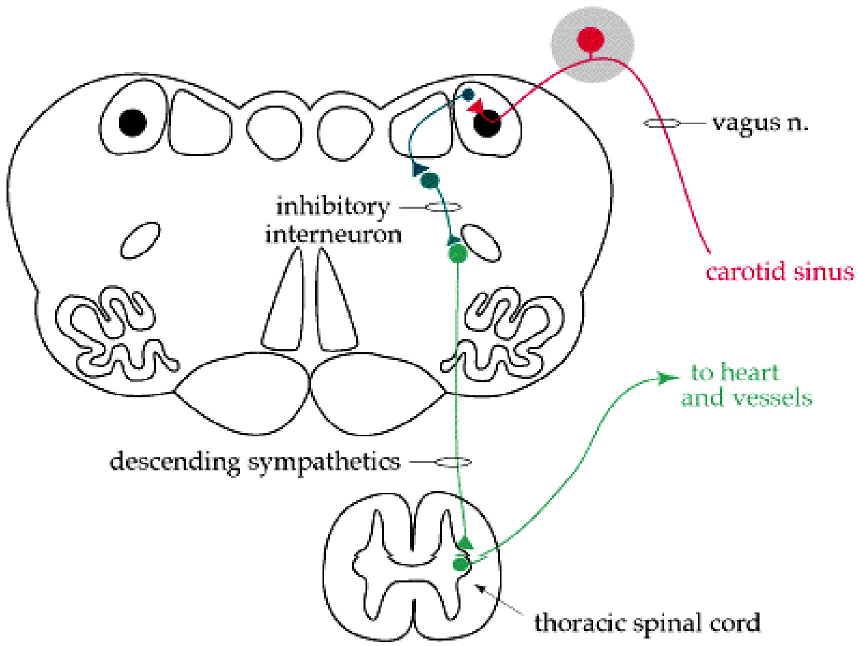
| | |
|---|---|
| pressure, carbon dioxide levels, gut distention, etc. | attack) is often referred to the shoulder. |

**D. The baroreceptor reflex:** A reflex is a pathway with an afferent signal (sensory) that evokes an efferent response (motor). The most common example is the stretch reflex, or knee-jerk reflex. A quick stretch of the tendon causes a brief contraction of the muscle. The autonomic system has several similar reflexes. One of these is the baroreceptor reflex, which maintains a constant blood pressure despite standing up or lying down.




The afferent signal comes from baroreceptors in the carotid sinus, a swelling of the carotid artery in the neck. If blood pressure suddenly jumps up, the baroreceptors respond and send the signal back to the nucleus of the solitary tract (NTS). Neurons in the NTS project to an adjacent vagal nucleus, the nucleus ambiguus, and excite the neurons that project to the heart. These acetylcholinergic neurons slow the heart, bringing down the blood pressure a little.

However, there is more to the story. In the knee-jerk reflex, for the quadriceps muscle to contract briefly, the hamstring muscle must also relax briefly. As a flexor-extensor pair, they must always receive opposite signals. The sympathetic and parasympathetic systems are like a flexor-extensor pair, so when activating the parasympathetic you must inhibit the sympathetic. Just like in the spinal cord, this is accomplished by an inhibitory interneuron.




When the high blood pressure signal arrives at the NTS, an inhibitory interneuron projects to the group of cells that control the sympathetic neurons in thoracic cord. These cells are called the **descending sympathetics**. An important feature of the descending sympathetics is that they are constantly firing at a steady level. This enables them to be turned down - if a neuron was already silent, an inhibitory signal would make no difference. Therefore, in response to the surge in blood pressure, the descending sympathetics are inhibited, and the sympathetics in the spinal cord fire at a much lower rate. As a result, the heart and the blood vessels are allowed to relax, the heart slows, vasodilation occurs, and blood pressure drops. The inhibition of the sympathetic system is actually a more powerful way to lower blood pressure than activating the parasympathetic system.

**RETURN TO HOME PAGE**

# Final Year Undergraduate Projects 2003 - 2004

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### Dr. J.A.Illingworth:

- Chaotic behaviour during gene expression
- Computer modelling of mitochondrial metabolism
- Dilated cardiomyopathy
- Inflammation & ageing
- Nucleoside diphosphate kinase
- Tropical diseases

This page provides more detailed information on the final year undergraduate research projects available under my supervision during 2003 - 2004.

## Chaotic behaviour during gene expression: (computer project)

The negative feedback systems regulating gene expression have all the physical and mathematical features which can give rise to chaotic behaviour in other physical systems. It is hypothesised that chaotic gene expression would not normally offer any biological advantages, and that mechanisms will probably have evolved to prevent it happening.

Theoretical arguments suggest that chaos could be best avoided by reducing the sensitivity of the feedback loops, and by reducing any delays between the sensing mechanism and the biological response. These have implications for the "design" and evolution of biochemical control mechanisms. It may be possible to prove that the mixture of allosteric, covalent modification and genetic control systems observed in many organisms should outperform any of the three mechanisms used in isolation. It seems possible that the attentuation systems often associated with polycistronic bacterial mRNA could be particularly effective in avoiding chaotic behaviour.

Students will use computer model building to test these hypotheses [either Excel spreadsheets or direct coding in C++] and construct model systems lacking one or more of the normal biological features to see how well these are expected to behave.

## Mitochondrial metabolism: (computer project)

Many of the technical problems previously associated with metabolic computer models have been solved in recent years. However, a whole cell is still a very large and complex system, and it is difficult to assemble a sufficient quantity of reliable metabolic data to properly distinguish between alternative models. This project will focus on the mitochondrial compartment, which is much smaller and simpler than a complete cell, so there is a greater prospect of useful results within a reasonable period of time.

## Dilated cardiomyopathy: (library project)

Dilated cardiomyopathy (DCM) is a severe, debilitating disease which is the most common indication for cardiac transplantation. It is characterised by huge increases in cardiac chamber size, ventricular wall thinning, and a loss of cardiac contractility. The dilatation may give rise to heart valve incompetence, further reducing the pumping efficiency of these diseased hearts. Systemic and pulmonary venous blood pressures are elevated, leading to fluid oedema in the lower limbs and to respiratory difficulties, which may be compounded by deep vein thrombosis and pulmonary embolism.

A bewildering variety of defects and insults give rise to the same end-stage pathology. A substantial minority of cases have an obvious genetic contribution, with identified mutations in cardiac structural proteins. For the majority of cases the precise cause remains unknown. This unpleasant disease is sometimes preceded by a Coxsackie virus infection, and may include an auto-immune component. Suggested auto-antigens include the mitochondrial adenine nucleotide carrier, cardiac myosin and the cardiac catecholamine receptor. However, in many patients it is difficult to demonstrate any tissue inflammation or immune response.

There is no known cause for the unwanted cardiac re-modelling which is such a conspicuous feature of this disease. A suggested thrust for this library project is to look at the mechanisms responsible for morphogenesis in normal hearts, in order to understand how these might give rise to the re-modelling seen in diseased hearts.

## Inflammation & ageing: (library project)

Inflammation has long been recognised as an important physiological process, particularly in response to infection and traumatic injury. Recently there has been a growing recognition that inflammation and the pro-inflammatory cytokines IL-1 IL-6 and TNF-α play major pathological roles in a much wider range of conditions, including the so-called "diseases of civilisation". Inappropriate inflammation is a major cause of the death and ill-health in all areas of the world, and attempts to control inflammation are a major reason for drug treatment. It is arguable that an aberrant inflammatory response is the dominant pathological process in the human body.

It is possible to get some idea of current research activity in these areas by counting "Web of Science" hits for each disease between January 2002 and March 2003. Some results are tabulated below:

| pathology | total publications Jan 02 to Mar 03 | papers mention inflammation | percent |
|---|---|---|---|
| Alzheimer’s disease | 4875 | 311 | 6.4 |
| arthritis | 6484 | 1863 | 28.7 |
| asthma | 5129 | 1465 | 28.6 |
| atherosclerosis | 5043 | 898 | 17.8 |
| cachexia | 271 | 58 | 21.4 |
| cancer | 47861 | 1535 | 3.2 |
| cardiomyopathy | 2617 | 140 | 5.3 |

| | | | |
|---|---|---|---|
| coeliac disease | 668 | 97 | 14.5 |
| cholecystitis | 222 | 24 | 10.8 |
| Crohn’s disease | 1252 | 739 | 59.0 |
| diabetes | 12224 | 547 | 4.5 |
| emphysema | 500 | 87 | 17.4 |
| glomerulonephritis | 1065 | 174 | 16.3 |
| heart failure | 4946 | 225 | 4.5 |
| hypertension | 10125 | 414 | 4.1 |
| multiple sclerosis | 2563 | 574 | 22.4 |
| obesity | 4752 | 169 | 3.6 |
| pancreatitis | 1549 | 241 | 15.6 |
| pre-eclampsia | 264 | 12 | 4.5 |
| hemorrhagic fever | 265 | 9 | 3.4 |
| hyperthyroid | 545 | 16 | 2.9 |
| hypothyroid | 799 | 23 | 2.9 |
| vasculitis | 1047 | 254 | 24.3 |
| **Totals** | **115066** | **9875** | **8.6** |

Students will initially review the molecular biology of the "classical" inflammatory response, before examining recent research on inappropriate responses that may form part of the ageing process.

## Nucleoside diphosphate kinase: (laboratory project)

It is well established that ATP has a different phosphorylation potential in different parts of the same cell. This situation arises because the concentrations of [ATP], [ADP] and [Pi] differ in the various cell compartments. As a result the reaction

$$ADP + Pi = ATP + H_20$$

is further from equilibrium (i.e. more negative delta G for ATP hydrolysis) in some cell compartments than in others.

Metabolite transport across sub-cellular membranes makes a significant contribution to these effects. ATP is initially manufactured at a relatively low phosphorylation potential (i.e. at a lowish ATP:ADP ratio) by ATP synthase within the mitochondrial matrix space. It is subsequently transported into the cytosol by the adenine nucleotide carrier, which swops an $ATP^{4-}$ for an $ADP^{3-}$ and is consqently driven in the direction of ATP export by the mitochondrial membrane potential. This means that the cytosol has a higher ATP:ADP than the mitochondria, and ATP is "worth more" in the cytosol.

Until recently it was assumed that all the other nucleoside triphosphates had more or less the same phosphorylation potential as ATP, and that all were ultimately derrived from ATP through reactions catalysed by nucleoside mono- and di-phosphate kinases. However, there are reasons to doubt this simplistic view.

GTP is manufactured during the TCA cycle by the enzyme succinate thiokinase (STK) which catalyses the reaction:

succinyl-CoA + Pi + GDP = succinate + GTP + CoASH

A nucleoside diphosphate kinase (NDK) is required to transfer the high-energy phosphate to ADP in the reaction:

GTP + ADP = GDP + ATP

Unfortunately, the STK is in the mitochondrial matrix space, but it seems likely that NDK is in the intermembrane space, with no direct communication between the two enzymes. In addition, there are other GTP-linked enzymes in the matrix space, such as PEP-carboxykinase, fatty acid activation and a nucleoside MONOphosphate kinase:

GTP + AMP = GDP + ADP

which might suggest that the intramitochondrial GTP and ATP pools are kept apart, and that GTP has a higher phosphorylation potential than ATP inside the matrix space. This could be important for successful gluconoegenesis under conditions of metabolic stress (running for your life, for instance). The reaction sequence

ATP + fatty acid + CoASH = fatty acyl CoA + PiPi + AMP
fatty acyl CoA + GDP + Pi = fatty acid + GTP + CoASH

could behave as a "metabolic transformer" converting two low grade ATP-type phosphoanhydride bonds into 1 high potential GTP-type phosphoanhydride bond suitable for the manufacture of PEP, which is a much better phosphate donor than ATP.

The general strategy would be (1) to get the various enzyme assays working, and confirm the suspected position [pedestrian, but gets results in the bag early on] then (2) show by immunoblot or otherwise that the intra-and extra-mitochondrial forms of NDK and PEPCK really are different (either different genes or different splicing) and that there is not a dual intracellular location, and (3) look for evidence of compartmentation in mitochondrial nucleotide metabolism.

## Tropical Diseases: (library project)

This is a huge field and there are many diseases to choose from. I suggest you pick out one, or a closely related group. In recent years students have prepared reports on various aspects of Malaria, Chagas disease, Leishmaniasis, Schistosomiasis, Sleeping Sickness and Dengue fever but progress is now so rapid that it would be possible to pick these topics again. There is a lot of basic information in the Leeds Medical School Library, but if you want a quick overview of the various diseases, try the WHO website or the CDC website in the USA.

There is a really excellent account of African Trypanosomiasis by Moore et al (2002) NEJM **346**, 2069-2076. You might be asked to log in to this journal website, in which case the Health Sciences Library can supply a password.

There are several parasitology sites on the web, including (for example) David Gibson's website and a useful introduction from Leicester.

Many of these diseases have several things in common. The parasites are often protozoal, except for tuberculosis, leprosy and viral diseases! They often show antigenic variation, or steal surface antigens from their hosts. This may lead to autoimmune complications. Several parasite and vector genomes have been sequenced and this has suggested new drug targets which exploit differences between parasite and host metabolism.

| *Personal Interests* | *Home Page* | *Top of page* | *Research Interests* |
|---|---|---|---|

On older browsers, this might take a while because of the applet loading.

# MEMORY HIERARCHY DESIGN

Memory heirarchy design is based on three important principles:

- **Make the Common Case Fast**
- **Principle of Locality**
- **Smaller is Faster**

**These are the levels in a typical memory hierarchy.** Moving farther away from the CPU, the memory in the level becomes larger and slower .

The above principles suggest that we should try to keep recently accessed items in the fastest memory.

Because the smaller memories are more expensive and faster, we want to use smaller memories to try to hold the most recently accessed items close to the CPU and successively larger (and slower, and less expensive) memories as we move away from the CPU. This type of organization is called **a memory hierarchy** . Two important levels of the memory hierarchy are **the cache** and **virtual memory.**

To evaluate the effectiveness of the memory hierarchy we can use the formula:

Memory_stall_cycles = IC * Mem_Refs * Miss_Rate * Miss_Penalty

| | | |
|---|---|---|
| where | IC | = Instruction count |
| | Mem_Refs | = Memory References per Instruction |
| | Miss_Rate | = the fraction of accesses that are not in the cache |
| | Miss_Penalty | = the additional time to service the miss |

# Tutorial 3: DC Machines

1) A 2kW series DC motor has an armature and field resistance of 0.16 ohms. If the power output from the motor = 1600W at 1,200rpm, with an armature current of 25A and a supply voltage of 72V, calculate: 1) the torque and emf constants for this motor; 2) if the supply has a maximum power rating of 2000W, calculate the maximum torque available from the motor when taking the maximum power available from the supply. 3) Calculate the efficiency of the motor at this maximum value of torque 4) Calculate the new speed of this motor if the motor torque is reduced by 80%

Ans: 1) Kt = 0.02 $Nm/A^2$, Ke = 0.02 V/A.rad/s, 2) Tmax = 15.4 Nm, 3) eff = 88%, 4) 2619rpm

2) A DC shunt motor has the following parameters: Ra = 0.16 ohms; Rf = 36 ohms; supply voltage of 72V; Kt = 0.278 Nm/A2; torque of 12.73Nm at a speed of 1200rpm. 1) Calculate the emf constant, armature current and efficiency of this motor at this value of load. 2) What is the new speed of the motor if the torque is reduced by 80%? 3) Calculate the field resistance required to increase the speed of the motor to 2000rpm with a load torque of 12.73Nm.

Ans: 1) Ke = 0.272 v/A.rad/s, $I_A$=22.9A, eff=89%, 2) speed = 1251 rpm, 3) Field resistance = 62.6Ω

3) An electric vehicle requires a starting torque of 300Nm to accelerate at an appropriate rate from rest. The battery voltage of the vehicle is 240VDC and the motor is a series type.
   From the following information derive the motor parameters and the torque and emf constants:
   a) At start up the motor takes a current of 120A.
      At a speed of 1,500rpm the torque is 42 Nm.
   b) Calculate the efficiency of this motor at 500rpm and at 1500rpm.

Ans: a) $(R_a + R_f) = 2\Omega$, $K_T = 0.02Nm/A^2$, $K_e = 0.021V/(A.rad/s)$, b) $eff_{500} = 33.7\%$, $eff_{1500} = 60.7\%$

4) If a shunt connected DC motor replaces the series motor in question 3) we still require to achieve 300Nm of torque to start the vehicle from rest.
   Calculate the motor torque and emf constants if:
   a) $R_f = 24\Omega$ and $I_a = 600A$ at start up and $I_a = 469A$ at 500rpm.
   b) What torque is available from this motor at 1500rpm?
   c) Calculate the efficiency of this motor at 500rpm and at 1500rpm and compare your answers with those from question 3b). Which motor is better suited for an electric vehicle application. Explain your answer.
   d) If the torque is halved from that at 1500rpm (i.e. the value from part b)) calculate the new value of armature current required at this speed. Further if the motor is now to be run at 2000rpm at this new value of torque, calculate the new value of field resistance required to achieve a speed of 2000rpm. By varying the field resistance we are 'field weakening' the motor.

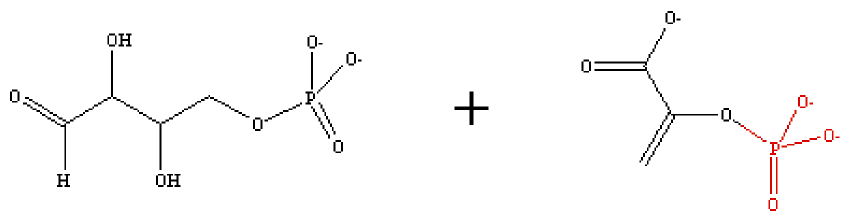
Ans: a) $K_T = 0.05Nm/A^2$, $K_e = 0.1V/(A.rad/s)$, b) T = 103.65Nm, c) $eff_{500} = 10.7\%$, $eff_{1500} = 31.2\%$, d) $I_a = 103.65A$, $R_f = 25.7\Omega$.

# Shikimate Pathway

The 3-D version of this pathway doesn't yet exist.

## Overall Reaction for the Pathway

D-Erythrose-4-phosphate + 2 Phosphoenoylpyruvate + $NAD^+$ + NADPH + ATP ---> Chorismate + NADH + $NADP^+$ + ADP + 4 $P_i$

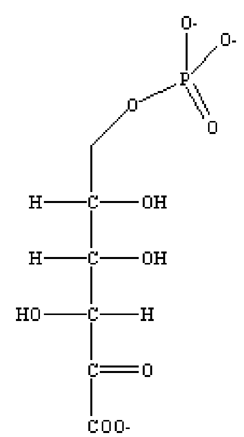



D-Erythrose-4-Phosphate + Phosphoenoylpyruvate

$H_2O$

$P_i$

Enzyme Complex



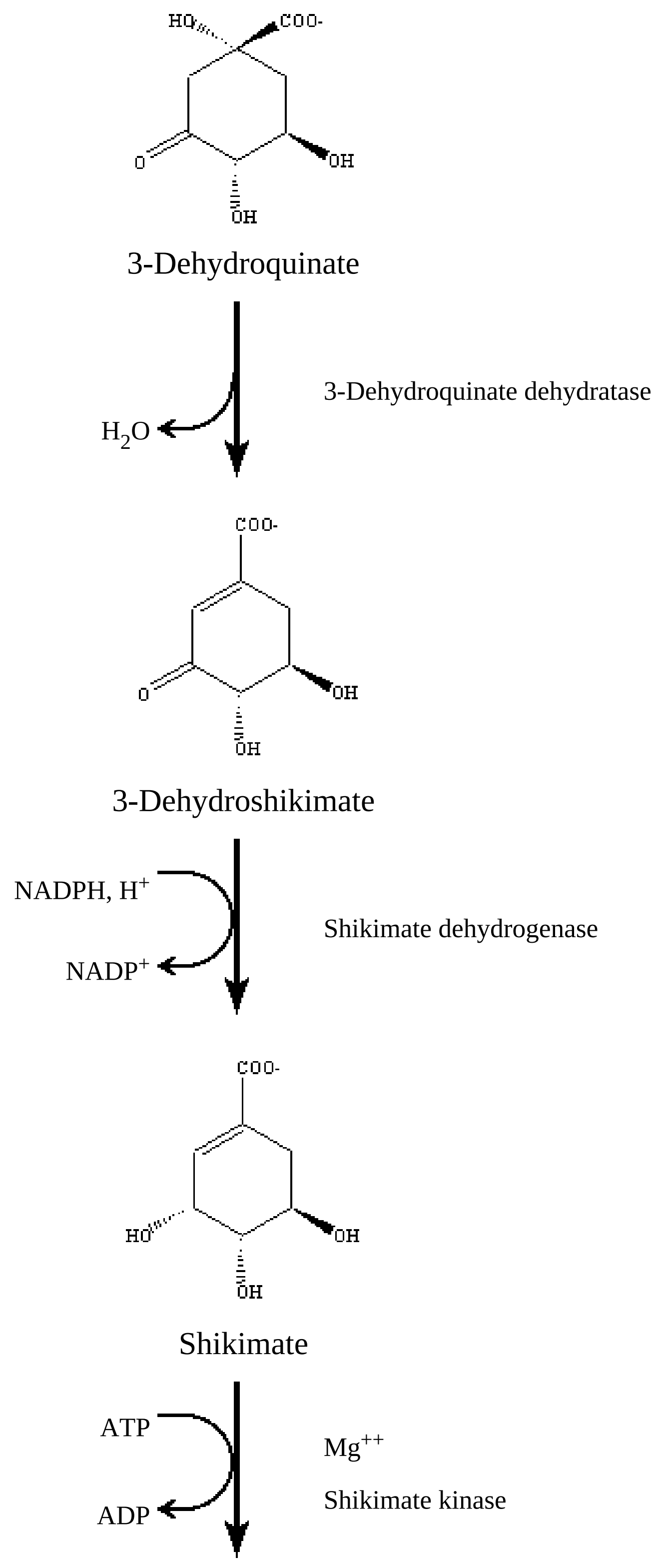

7-Phospho-2-dehydro-3-deoxy-D-arabinoheptulosonate

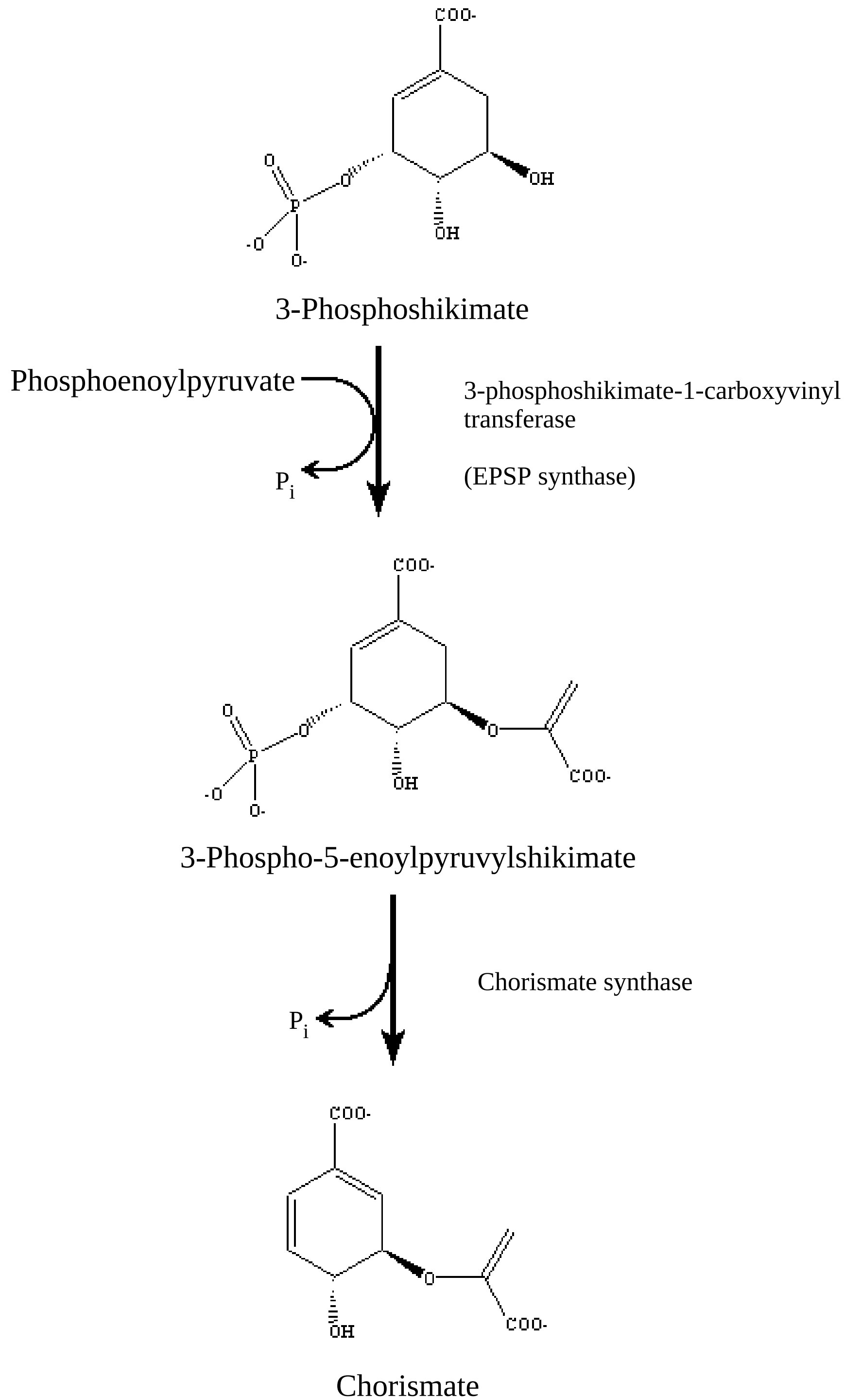
$NAD^+$

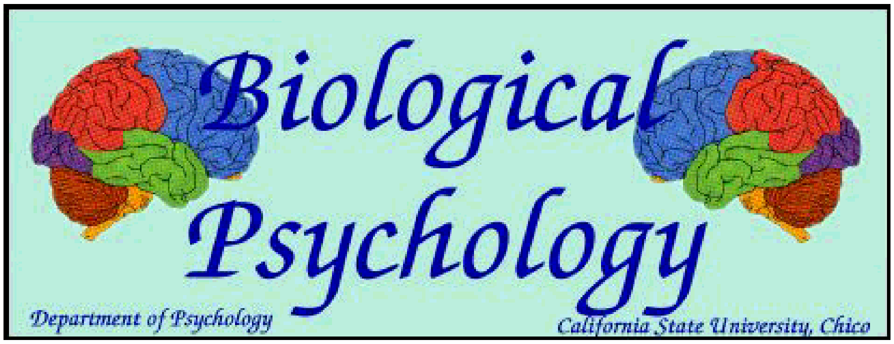
NADH, $H^+$, $P_i$

$Co^{++}$

3-Dehydroquinate synthase


HO
COO-
O
OH
OH
3-Dehydroquinate
3-Dehydroquinate dehydratase
$H_2O$
COO-
O
OH
OH
3-Dehydroshikimate
NADPH, $H^+$
$NADP^+$
Shikimate dehydrogenase
COO-
HO
OH
OH
Shikimate
ATP
ADP
$Mg^{++}$
Shikimate kinase


COO-
O
P
O
-O
O-
OH
OH
3-Phosphoshikimate
Phosphoenoylpyruvate
Pi
3-phosphoshikimate-1-carboxyvinyl transferase
(EPSP synthase)
3-Phospho-5-enoylpyruvylshikimate
Chorismate synthase
Pi
Chorismate




# Neurotransmission

**[BioPsyHome]**

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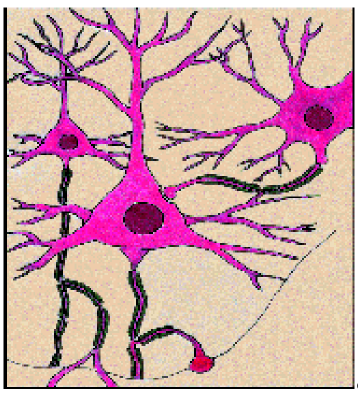

## . . .Some Brain Facts.

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. . .The brain is made of cells (called neurons) which communicatewith each other at places called synapses. A synapse is a functional (but not physical) contact between two neurons. There are about 100 billion neurons in the human brain and each has about 10,000 contacts with other neurons. The number of synapses in the human brain is about 10 to the 15th power. The neurons of one human cerebral cortex would reach over 250,000 miles if placed end to end. The complexity of this organ is obviously enormous!

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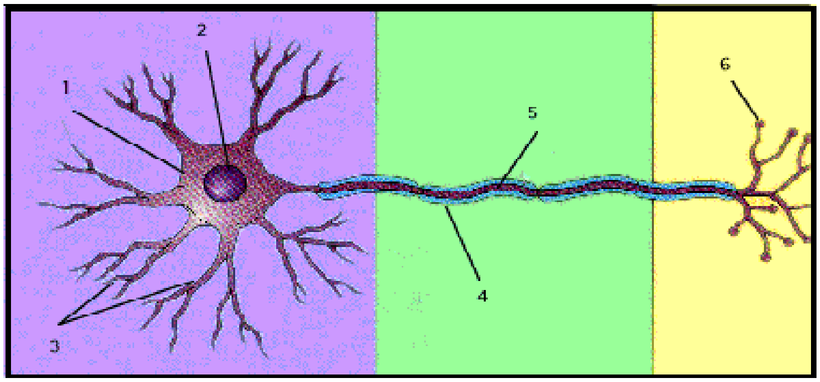
## Do You Know the Parts of a Neuron?

### Test Yourself - Click on an area for the answers.

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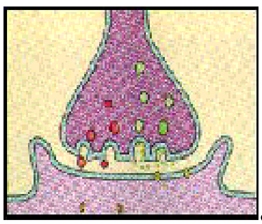


## . . . How does one neuron communicate with another neuron?

. . . . .Neurons generate electrical events called action potentials which consist of brief reversals in the polarity (electrical state) of the axon (transmitting region) of the cell. These action potentials cause the release of a chemical messenger from a storage vesicle in the axon terminal. The chemical messenger (called a neurotransmitter) travels across a synapse to bind to a postsynaptic receptor protein. The act of binding to the receptor protein sets in motion a series of events which eventually brings about a change in the electrical state of the postsynaptic cell. Some neurotransmitter-receptor bindings excite the cell and others inhibit it. At any given moment a neuron receives thousands of these messages and integrates this input to bring about only one of two possible outcomes - the neuron stays in a resting state or it generates an action potential to communicate with another neuron.

.

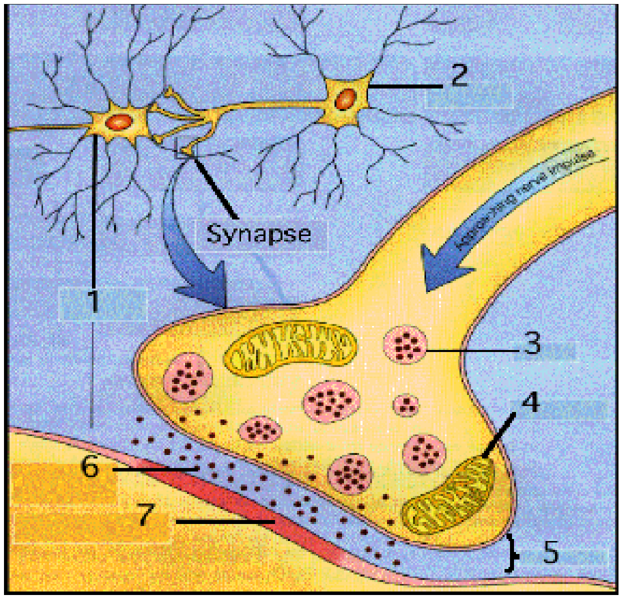
[**Question**: If each neuron of the brain (approximately 100 billion) can be in only one of two states (resting or acting), then what would be the number of functionally different brain configurations possible?] See answer below.**

.

# The Synapse

## Can You Identify the Parts of a Synapse?

### Click on the area for the answers.

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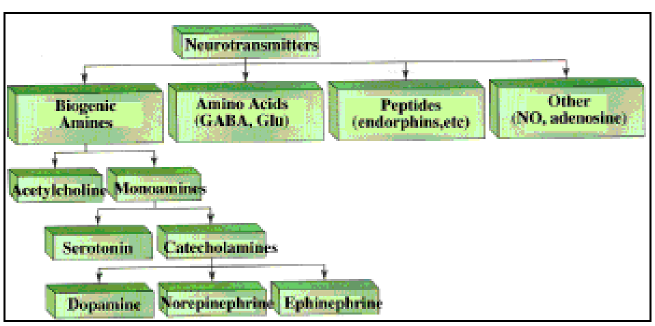
# Neurotransmitters

.....Neurotransmitters are chemicals which are released into the synaptic space whenever a neuron conducts an action potential to the axon terminals. There are perhaps 100 or so different neurotransmitter varieties in the brain. Some of them are shown in the flow chart below. Each neurotransmitter plays some role in most behaviors, but often we identify a key behavior with each neurotransmitter.

### Can You Identify Some of the Key Behaviors Associated with Each of these Neurotransmitters?

### Click on the boxes for the answers.

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**[2 raised to the 10 to the 11th power; according to Carl Sagan this is a number greater than the number of elementary particles (protons and electrons) in the universe.]

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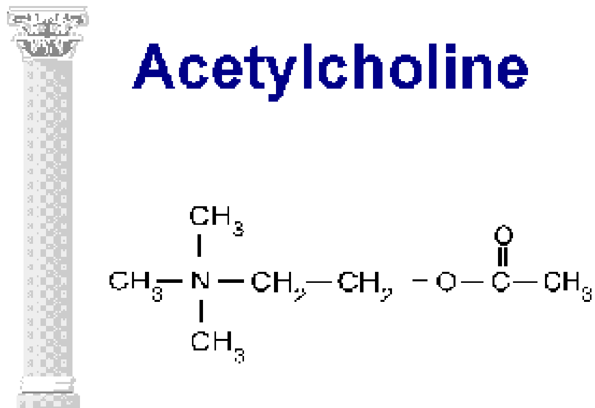
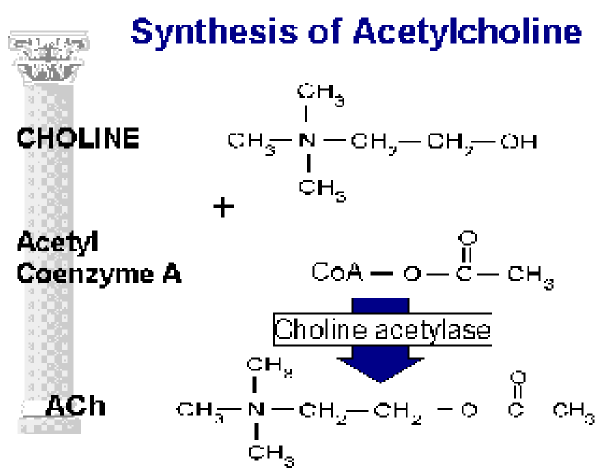
**[BioPsyHome]**

# Chemical Neurotransmitters




## *Acetylcholine*

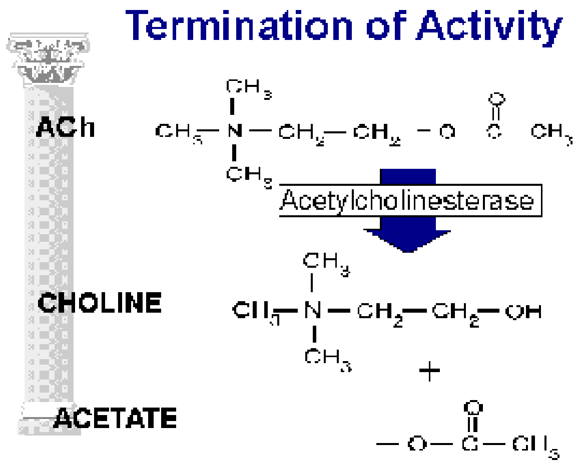
Acetylcholine is both a central and peripheral neurotransmitter. It is found diffusely throughout the CNS. CNS therapeutic applications include cholinometic approaches in the treatment of Alzheimer's disease and anticholinergic drugs in the treatment of Parkinson's disease. Acetylcholinesterase inhibitors and anticholiniergics are also used as reciprocal antidotes for one another. The former agents are also used as insecticides and in chemical warfare.

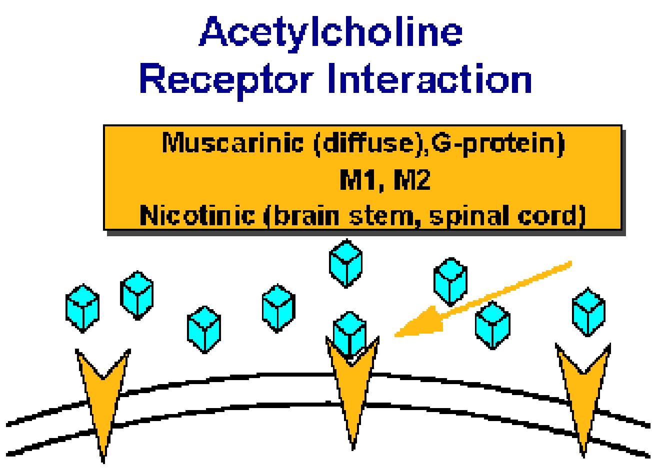



Acetylcholine is synthesized from choline and acetylCoA by the enzyme choline acetylase. Enzyme inhibitors such as hemicholinium deplete ACh and diminish cholinergic transmission.

The synaptic action of acetylcholine on the

The synaptic action of acetylcholine on the receptor is terminated by enzymatic cleavage by acetylcholinesterase. The choline resulting from the hydrolysis is taken back up into the neuron terminal for resynthesis of ACH.




There are both nicotinic and muscarinic receptors in the CNS, so named because of selective sensitivity to the two alkaloids. The muscarinic receptors are the most diffuse and are G-protein coupled; a $M_1$ receptor is excitatory and a $M_2$ receptor is inhibitory. Nicotinic receptors are found more selectively in the brainstem and spinal cord (e.g., the Renshaw cell).

Return to index: Drugs, Transmitters, Dieseases, Enzymes, Areas Terms

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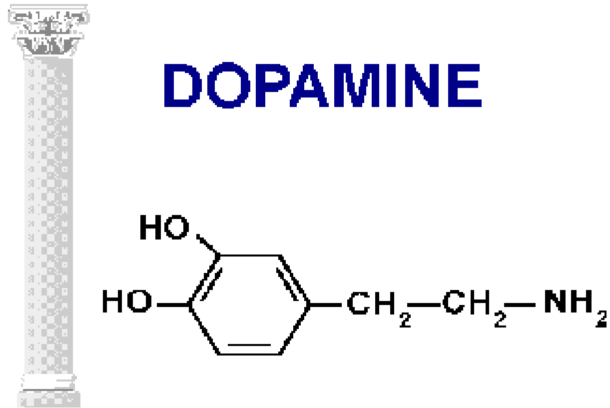
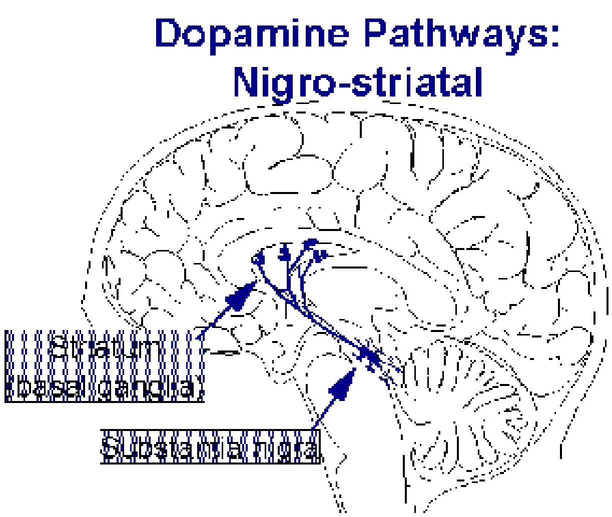
# *Dopamine*



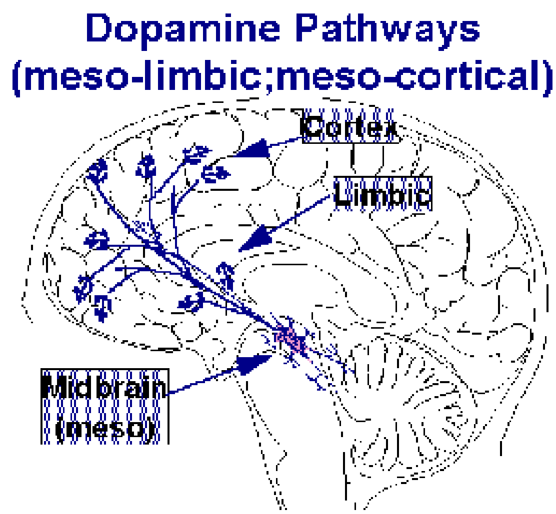

Dopamine is found in both small and large local circuit pathways in the CNS. The latter include the nigro-striatal pathways involved in the etiology of Parkinson’s Disease and meso-limbic/meso-cortical pathways implicated in pyschosis. L-dopa, which is converted to dopamine is the primary replacement therapy in Parkinson’s Disease. Dopamine antagonists are used in the treatment of psychosis. Dopaminergic input to the chemoreceptor trigger zone is the basis of the use of the agonist apomorphine as an emetic to treat poisoning, and the use of antagonists as antiemetics. Hypothalamic dopaminergic neurons inhibit prolactin secretion and lead to the use of agonists in inhibiting lactation. There are a number of drugs which interact presynaptically with dopamine terminals including reserpine, amphetamine, MAO inhibitors, and cocaine.

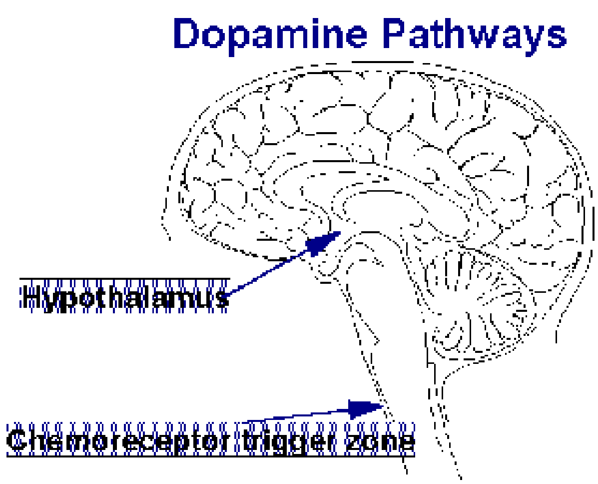



The nigro-striatal pathways consist of dopaminergic neurons which originate in the substantia nigra (black substance) in the midbrain and project to the striatum (basal ganglia), indirectly controlling involuntary motor movement. Parkinson’s disease results from degeneration of these neurons with depletion of dopamine from the neuron terminals and a loss of control of involuntary motor control (tremor, rigidity, etc) mediated by the extra-pyramidal pathways. This forms the basis for the administration of the dopamine precursor, l-dopa as replacement therapy, dopamine agonists, and drugs which indirectly facilitate dopamine activity.

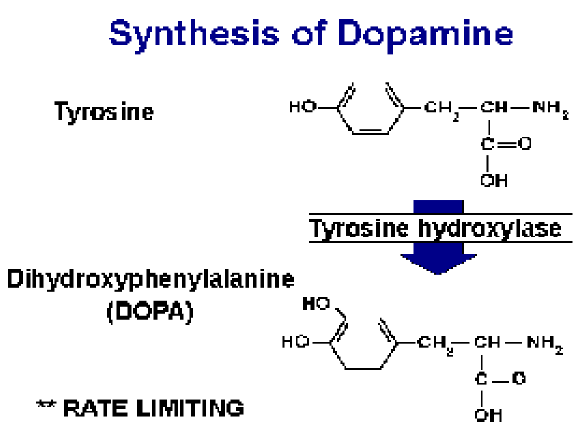
The dopaminergic meso-limbic and meso-cortical

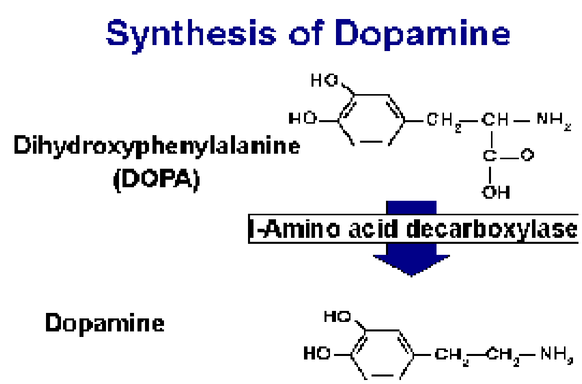
The dopaminergic meso-limbic and meso-cortical pathways arise in the midbrain (meso) and project to parts of the limbic system and cortex. Psychosis (e.g., schizophrenia, drug-induced) seems to reflect excess dopaminergic activity involving the $D_2$ family of receptors. This forms the basis for the antipsychotic action of drugs which are antagonists on these receptors or deplete dopamine from neuron terminals (reserpine).



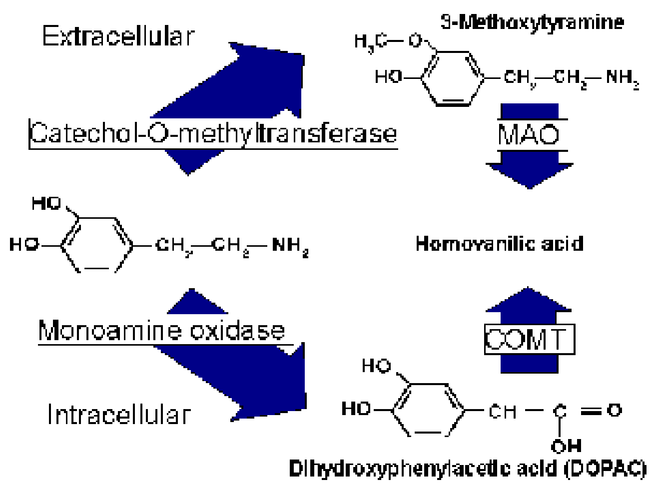




Dopaminergic neurons are found in the hypothalamus which increase the secretion corticotropin and growth hormone and inhibit the release of prolactin. This leads to a secondary use of l-dopa to inhibit lactation (prolactin), and to side effects of some dopamine antagonists which can produce some breast engorgement and decrease the cortisol response to stress. Despite the effects of l-dopa in increasing growth hormone, acromegaly does not occur. The chemoreceptor trigger zone (CTZ) is found in the medulla and responds to dopaminergic neuron and the $D_1$ agonist apomorphine inducing vomiting.



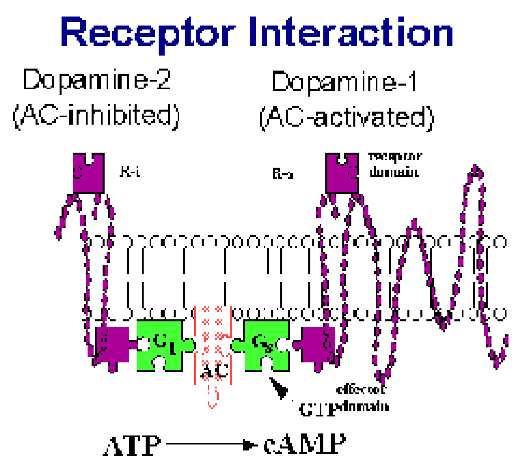



Dopamine is synthesized by a two step process from the l-amino acid tyrosine. The rate limiting step is the hydroxylation of tyrosine by tyrosine hydroxylase to the catechol di-hydroxyphenylalanine (DOPA). DOPA is then decarboxylated by the non-specific enzyme l-amino acid decarboxylase to dopamine. Although there are inhibitors of both enzymes, none are useful clinically.

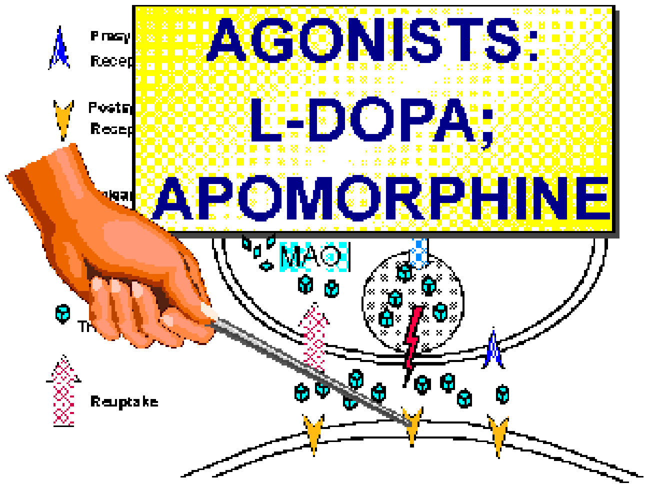



Dopamine is metabolized extracellularly by the enzyme catechol-o-methyl transferase. This reaction does not effect the termination of action of dopamine in the synapse, and there are no important inhibitors of the enzyme. Monoamine oxidase is found within mitochondria in the neuron terminal and serves to regulate intracellular stores of dopamine. This is a non-specific enzyme which oxidatively deaminates other monoamines such as norepinephrine and serotonin. Inhibitors of this enzyme increase the intraneuronal stores of all three monoamines. The increase in dopamine is the basis of the use of the MAOB inhibitor, selegiline in Parkinson's disease.

There are at least 5 dopamine receptors denoted by subscripts $D_1$-$D_5$. The $D_1$ and $D_5$ receptors (referred to as the $D_1$ family of receptors) act on G-protein coupled receptors which activate adenylate cyclase, increase cyclic AMP, and are excitatory. The $D_2$, $D_3$, and $D_4$ receptors (the $D_2$ family act on G-protein coupled receptors which inhibit adenylate cyclase, decrease C-AMP, and are inhibitory. This family of receptors are more concentrated in the meso-limbic and meso-cortical pathways and are associated with psychosis and its treatment. Genetic polymorphisms exist for the $D_4$ receptor which might explain some of the genetic basis for schizophrenia.

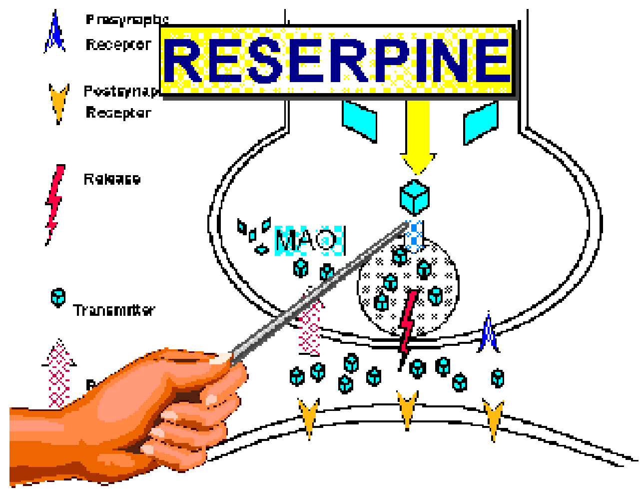



L-dopa is taken up into the neuron terminal and

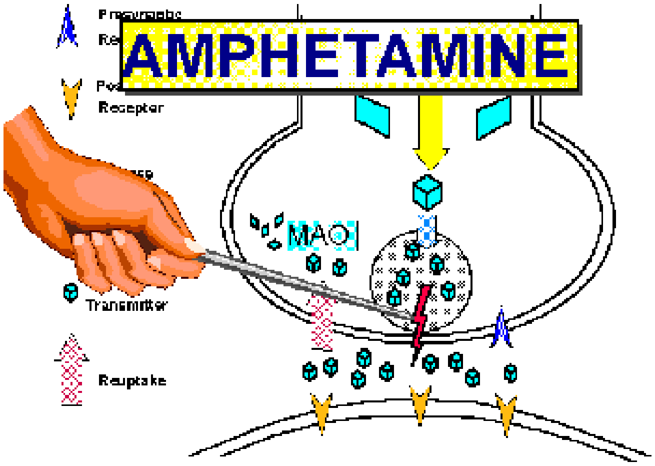



L-dopa is taken up into the neuron terminal and metabolized to dopamine, replenishing deficient stores in Parkinson's disease. Upon release, all dopamine receptors are affected, but it is unclear which receptor interaction is important in Parkinson's disease. L-dopa is also used to suppress action because of its hypothalamic action. Other agonists used in Parkinson's disease include the ergolines which are more selective for receptor subtypes. The dopamine agonist apomorphine is used as an emetic because of its dopaminergic action on the chemoreceptor trigger zone.

Reserpine blocks the intraneuronal storage of dopamine, serotonin, and norepinephrine both centrally and peripherally. With the active storage mechanism blocked, free transmitter is oxidized by mitochondrial MAO, and the terminals are markedly depleted of the neurotransmitters. This mechanism on dopamine was the basis for its now obsolete use in psychosis. The depletion of norepinephrine lowers blood pressure and forms the basis for the wanning use of this drug in hypertension. Depletion of norepinephrine and/or serotonin underlies the ability of the drug to produce mental depression, and this mechanism partially underlies the amine theory of affective disorders.

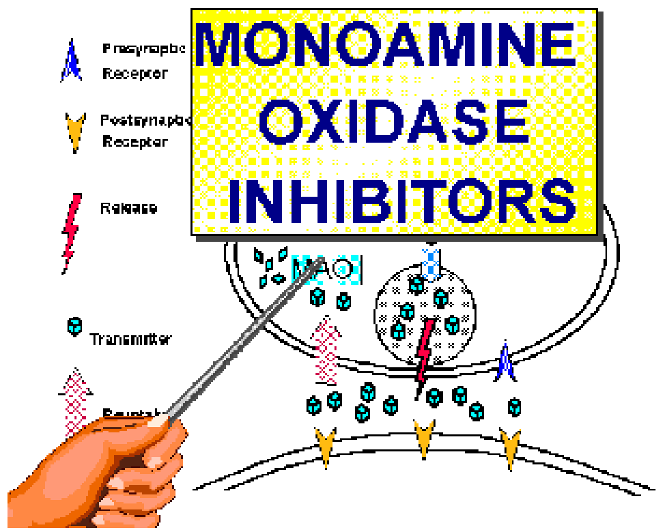



Amphetamine, and other so-called indirect-acting

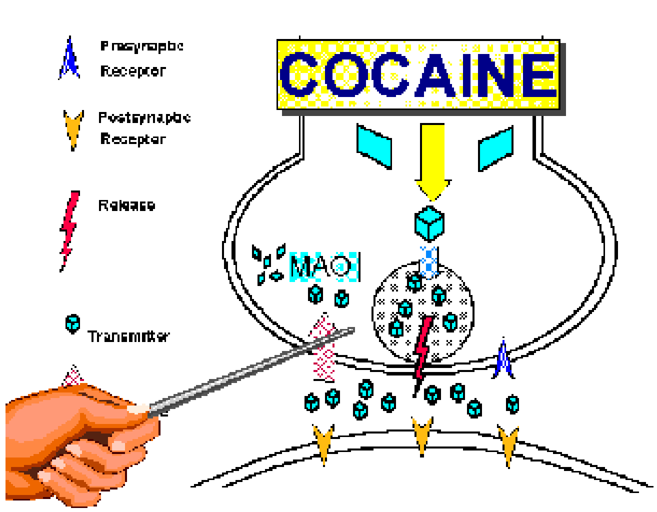



Amphetamine, and other so-called indirect-acting sympathomimetics, cause the release of dopamine and norepinephrine centrally as well as peripherally. The resulting effects of dopamine in the synapse explain the stimulant, euphoric and potentially psychologenic effects of the drug and are the basis for its use (and that of the preferred drug, methylphenidate) in attention deficit hyperactivity disorder. The effect in causing norepinephrine release is the basis of the anorectic effect of amphetamine and related drugs.

Mitochondrial monoamine oxidase (MAO) degrades intraneuronal dopamine, norepinephrine, and serotonin which is not protected by the storage vessicle. Inhibition of this enzyme results in a marked increase in the levels of these transmitters both centrally and peripherally. Increased levels of norepinephrine and/or serotonin are responsible for the antidepressant effects of non-specific MAO inhibitors such as tranylcypromine. The selective inhibition of the more selective is coenzyme MAOB by selegiline is the basis for its use in increasing dopamine stores in treating Parkinson’s disease. Inhibition of hepatic MAO and other enzymes by the non-selective agents is responsible for a large number of interactions which limit the use of these drugs.

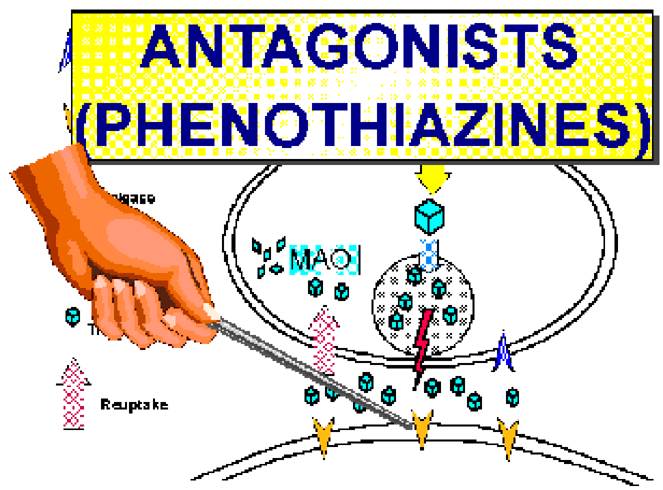



Cocaine inhibits the active uptake of the



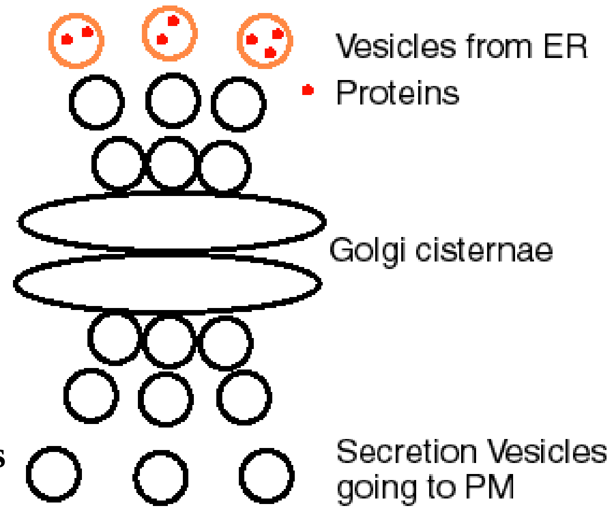

Cocaine inhibits the active uptake of the catecholamines dopamine and norepinephrine into neuron terminals thereby allowing accumulation of the transmitters in the synapse. The accumulation of dopamine, and its action on dopamine receptors, is responsible for the stimulant, euphoric, and possibly psychotogenic activity of the drug and its consequent abuse. The effect on norepinephrine results in anorectic and peripheral sympathomimetic effects.

Most dopamine antagonists, or neuroleptics (e.g. the phenothiazine, chlorpromazine)act non-selectively on most or all dopamine receptors; some of the newer ones are more selective (e.g., clozapine on the $D_1$ and $D_4$ receptor). The primary use of the agents in psychosis is based on dopamine antagonism in the meso-limbic and meso-cortical pathways. This action is based antagonism of dopamine on the $D_2$ family of receptors, perhaps more specifically the $D_4$ receptor. Actions in the nigro-striatal pathways by most, produce extrapyramidal side effects, and some minor indications. Antagonism in the chemoreceptor trigger zone is the basis for the antiemetic use of some of these agents.

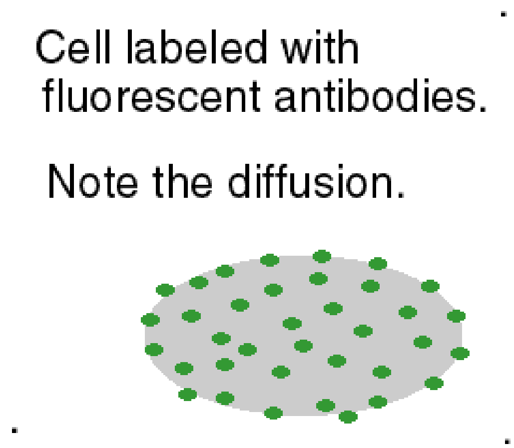



Return to index: Drugs, Transmitters, Diseases, Enzymes, Areas Terms

Shown to the right is an animation which illustrates how membrane flows through the stack of Golgi cisternae. Note throughout this process that the newly synthesized membrane "flows" along with the newly synthesized proteins. Membrane is synthesized by the combined action of the RER and SER. The RER synthesizes membrane proteins as well as proteins destined for secretion. The SER synthesizes membrane lipids. Thus, new memrane is synthesized and leaves the SER as vesicles which move on to the Golgi where they coelesce together at the forming face of the Golgi. The animation begins as the vesicles from the SER approach the Golgi. The membrane moves sequentially through the Golgi cisternae and leaves the Golgi at the maturing face as secretion vesicles bleb off. The SV move away from the Golgi and will later fuse with the plasma membrane (not shown here, but see the membrane fusion animation) and dump their contents (the soluble proteins synthesized in the RER) outside of the cell.



**Evidence for lateral diffusion of proteins**




**The cell is gray. The green dots illustrate the presence of fluorescently labeled antibodies which have bound to proteins on the surface of the cell. The green dots move across the surface of the cell. The researcher can apply a tiny but intense laser beam which bleaches the flourescent labels in a small area on the surface of the cell. This photobleaching produces a patch of unlabeled cell surface that appears gray. With time, however, proteins with fluorescent labels that were not bleached diffuse back into the area which was bleached and the bleached proteins diffuse away. The bleached spot disappears and the cell is uniformly labeled again. The rate at proteins diffuse back into the bleached area is much slower than is illustrated in this animation. This type of experiment has been used to provide evidence that proteins are free to diffuse laterally across the plane of the membrane. Similar experiments can be with fluorescently labeled lipids. In the case of lipids, the lateral diffusion rates are much greater.**

**Back to "Directory of Animations"**

**Back to Berg's Home Page**

# Dolby Surround Mixing Manual

Part No. 91536 Issue 2

Dolby

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This manual is dedicated to Bob “Spider” Seiderman, whose insight, experience and guidance through more than 20 years of audio mixing helped to make this manual possible. Without Bob’s input, the success of Dolby Surround for live sporting events would have been far more difficult. His expertise and willingness to share his thoughts with others are reflected in several sections of this manual. It is with deep sorrow that we have had to say our goodbye’s to Bob so early in his life.

Thank you Bob for letting us share some of your expertise with other audio engineers in the industry.

Your friends at Dolby Laboratories,

Jim Hilson
David Gray
Michael DiCosimo

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Part Number 91536 Issue 2

# Table of Contents

# List of Figures

# List of Tables

# Chapter 1
# Introduction

Dolby Surround is a format that enables the production and delivery of multi-dimensional soundtracks for television, cable, consumer video, compact disc, video game and other stereo media. Once created, Dolby Surround soundtracks can be recorded, broadcast and reproduced in the same manner as any conventional stereo programs, including compatible monophonic playback. Consumers equipped with Dolby Surround systems will experience a program's full measure of spatial dimensionality, just as they do from thousands of Dolby Stereo movies currently available on home video media.

Many aspects of Dolby Surround soundtrack production are the same as those of stereo soundtrack production. The main difference is that the mixing console must have at least three and preferably four outputs to feed the Dolby Surround encoder. To complete the surround system, additional speakers and amplifiers are needed to monitor the Center and Surround channels via a Dolby Surround decoder.

In most cases the finished two-channel encoded soundtrack is all that will be recorded or broadcast. However, in some cases it may be desirable to record the four-channel stems (Left, Center, Right and Surround encoder input signals) onto separate tracks when further elements are to be added later, such as with music pre-mixes for movie soundtracks.

This manual contains the information necessary for production personnel to properly produce soundtracks in Dolby Surround.

# Chapter 2
# Technical Guidelines

## 2.1 Equipment from Dolby Laboratories

The equipment described below exists in two versions. The newer one has much greater immunity to RF interference, and as sold in Europe with a single power line voltage, is compliant with the EMC standards of the European Union. Operationally the two versions are analogous, and this manual applies to both. Older units have a “gold” finish to the tray and top, the more recent ones a “silver” finish. The newer versions do not have removable front cover plates and thumbscrews; to gain access to the plug-in modules and internal switches, remove the extruded front panel (two screws on top and five underneath).

### 2.1.1 Dolby Model SEU4 Surround Encoding Unit

The SEU4 receives four input signals (Left, Center, Right, and Surround) from the audio console and matrix encodes them into two output signals (Lt and Rt). The Lt and Rt signals are then treated as any stereo signal would be for transmission and recording.

### 2.1.2 Dolby Model SDU4 Surround Decoding Unit

The SDU4 decodes the two-channel encoded signal (Lt and Rt) into four output signals (Left, Center, Right and Surround) using Dolby Surround Pro Logic decoding technology. The unit also provides switchable stereo and monophonic monitoring modes for evaluating mix compatibility. A ganged master fader allows all four monitor output channels to be varied together, allowing variations in listening level while maintaining playback balance and calibration.

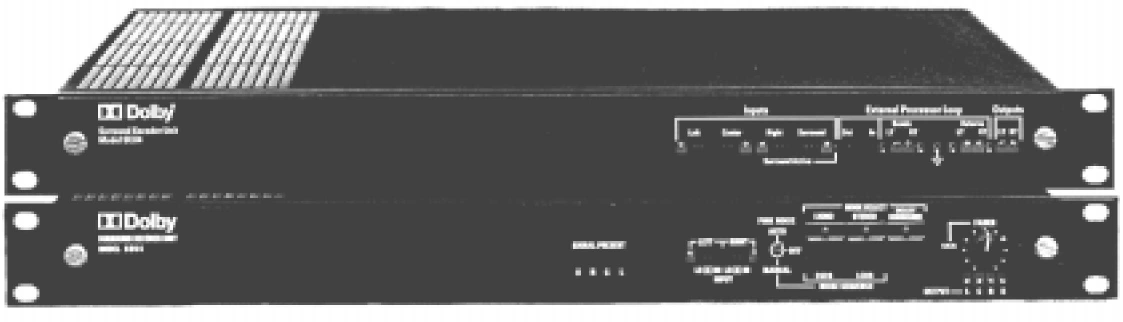

**Figure 2-1 Dolby Model SEU4 and SDU4**

It is important to listen through the decoder while mixing in order to hear any subtle changes that the Dolby Surround matrix encoding process may create.

Both the SEU4 and SDU4 are available for purchase from Dolby professional product dealers and for rent from several studio equipment rental houses.

## 2.2 System Information

### 2.2.1 Room Layouts

The various room possibilities for working with Dolby Surround encoding all conform to a basic standard. They all require the typical Left and Right speakers we have grown accustomed to for stereo production. In addition, a Center channel speaker and two or more Surround speakers are needed, *Figure 2-2*. Individual room requirements will determine exact needs for the Surround channel.

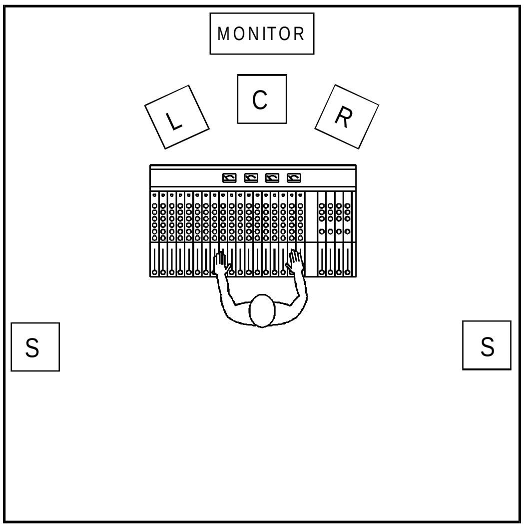


**Figure 2-2 Typical Room Layout**

### 2.2.2 Control Rooms

In order to achieve even Surround channel dispersion, Dolby Laboratories recommends that a room layout such as the one shown in *Figure 2-3* use two Surround speakers.

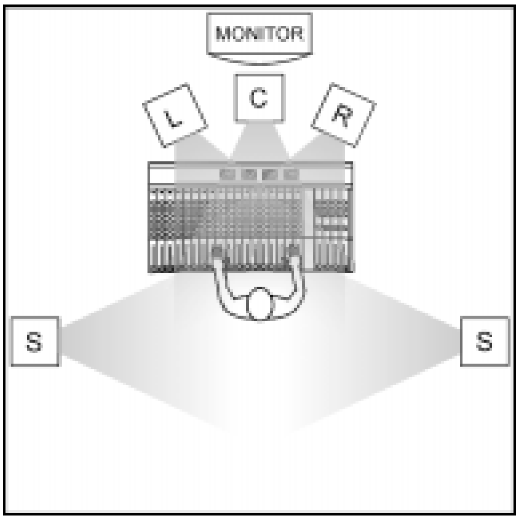


**Figure 2-3 Sound Field Pattern with Two Surround Speakers**

For rooms similar to *Figure 2-4,* four Surround speakers better serve both the mix engineer working in the front of the room and the clients listening in the back. Sharing two Surround speakers within this configuration compromises both listening positions: when the balance is correct for the engineer, it will usually be too loud for the client. When a compromise in balance is necessary, rooms should always be optimized in favor of the engineer. However, the better solution is for the engineer and client to each have a set of Surround speakers.

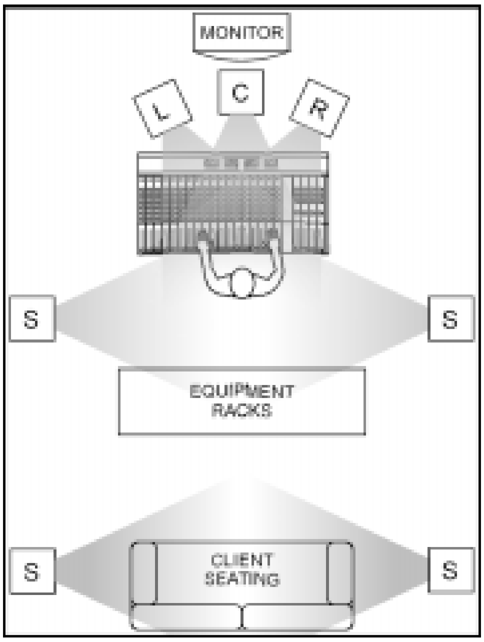


**Figure 2-4 Sound Field Pattern with Four Surround Speakers**

### 2.2.3 Remote Trucks

Remote trucks producing live shows offer a challenge for Center channel speaker placement. The desired Center channel speaker position is usually either occupied by equipment and video monitors vital to production, or by the window to the main production area. In the latter case a Center channel speaker would block the line of sight to the director. Neither scenario is desirable; the only reasonable solution is to not use a Center channel speaker. Because the mixer is in close proximity to the Left and Right speakers, successful mixes can be created using this arrangement. Since this is not an ideal situation, assistance from those at the station who can check the mix with a proper monitor setup is usually required.

### 2.2.4 Critical Listening Rooms

Critical listening rooms, mastering rooms, television master control rooms and screening rooms are similar to control rooms. For larger rooms, several Surround speakers may be used in an array much like a movie theater as shown in *Figure 2-5*. In these applications, the SDU4 is ideal. However, many facilities also set up separate Home Theater rooms with a typical living room atmosphere. A living room environment with a complete consumer system including a Dolby Surround Pro Logic receiver, VCR, laser disc player, CD player, and consumer grade speakers gives clients a good idea of how the project will translate when played back at home.

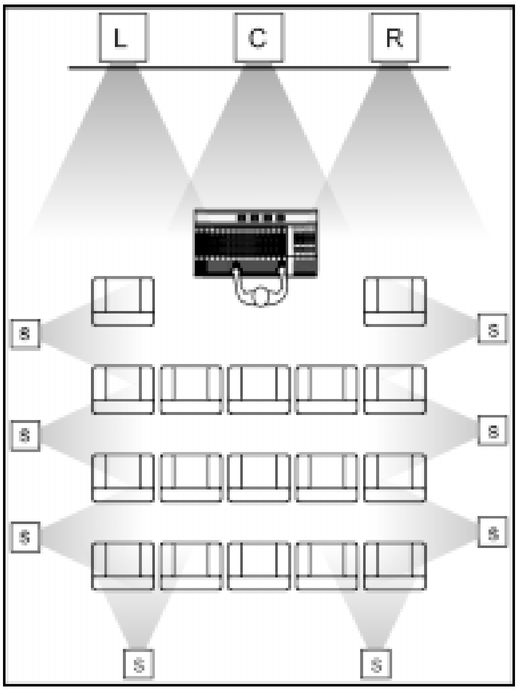


**Figure 2-5 Large Listening Room with Surround Speaker Array**

### 2.2.5 Consumer Decoders

Consumer Dolby Surround Pro Logic decoders operate identically to the Dolby model SDU4 professional decoder, but they include foolproof circuitry such as auto-balance to correct left/right balance errors. When mixing or checking quality, this function is undesirable, since it can hide the very problems being checked for. These types of features are needed at the consumer level to correct channel imbalances introduced during transmission or tape duplication. Also, consumer decoders do not normally have enough time delay for studios or a simple way to compare Mono, Stereo and Surround compatibility. While suitable for small listening rooms, never use consumer decoders in control rooms as part of the mixing process.

## 2.3 Additional Equipment Required

### 2.3.1 Speakers and Amplifiers

Front speaker setup may be accomplished two ways. One, add a Center speaker that matches the acoustic characteristics of the existing Left and Right soffit speakers or two, install three identical near-field monitors. In either case, the design of all three front speakers must be identical; panning from one type of speaker to another causes great differences in the sound. This does not mean that they all have to be the same size. Large Left and Right speakers and a smaller Center speaker from the same product line are acceptable. If possible, the Center speaker should have the same high- and mid-frequency drivers as the Left and Right speakers.

When placing the speakers, all three front speakers should be equidistant from the mixing position, as shown in *Figure 2-6*.

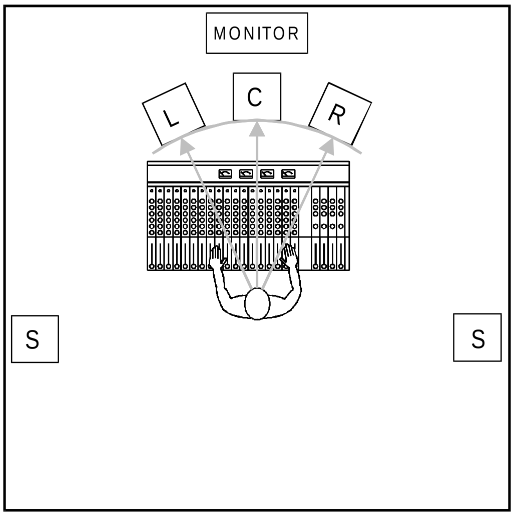


**Figure 2-6 Front Speakers Equidistant from Engineer**

Do not use soffit-mounted Left and Right speakers with a Center speaker placed on the console overbridge, as shown in *Figure 2-7*, because the Left and Right speakers will be too far from the mixer.

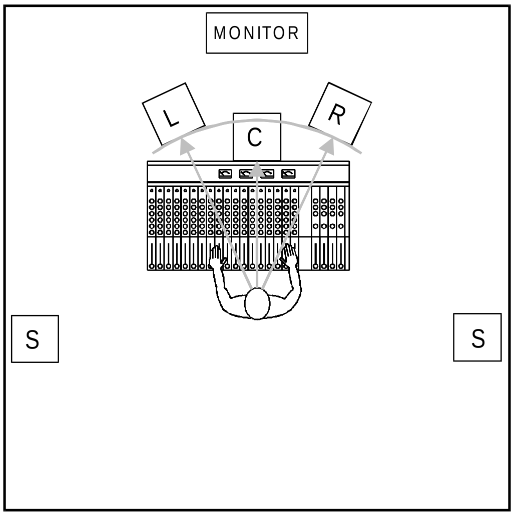


**Figure 2-7 Incorrect Soffit and Center Speaker Placement**

The Surround speakers can be smaller bookshelf-type speakers. The actual frequency response of the Surround channel is 100 Hz to 7 kHz so large speakers for bass reproduction and extended range tweeters for ultra-high frequencies are not necessary. It is important, however, to choose Surround speakers that sound similar to the front speakers throughout the 100 Hz to 7 kHz range; a smaller speaker from the same product line usually works best .

If all three speakers in the front are identical, the power amps for each should be rated equally. If the Center speaker is smaller, and the Center channel bass is being redirected to the Left and Right channels, see *Part 2.3.3,* then the power rating of the center amp should be at least 75% that of the left and right amps. The total power provided for the Surround channel should not be less than that of either the Left or Right channels. If separate amps are used for each Surround speaker (the preferred method), each amp should have at least 50% of the power of the left and right amps. If one amp is used for the Surround channel (acceptable, but not as desirable), it should be rated the same as left and right amps. For example, use three identical front speakers with three 100-Watt amps and two 50-Watt amps for the Surround speakers.

### 2.3.2 Center Channel Speaker

The Center channel speaker is used to anchor dialog and other sounds to the screen. In conventional two-speaker configurations, the listener can only hear a balanced mix when seated exactly in the center or sweet spot, *Figure 2-8*. This configuration provides a good phantom image.

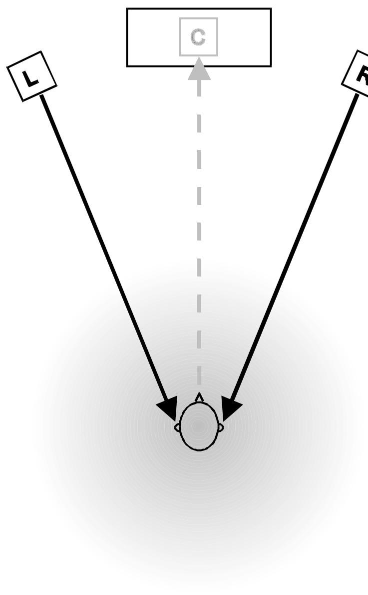


**Figure 2-8 Listener in Sweet Spot**

If the listener moves to either side of this sweet spot the mix becomes heavy on that side. The listener perceives the Center channel as coming from a point other than halfway between the Left and Right speakers, as in *Figure 2-9*. In this configuration, the phantom image is displaced off the screen.

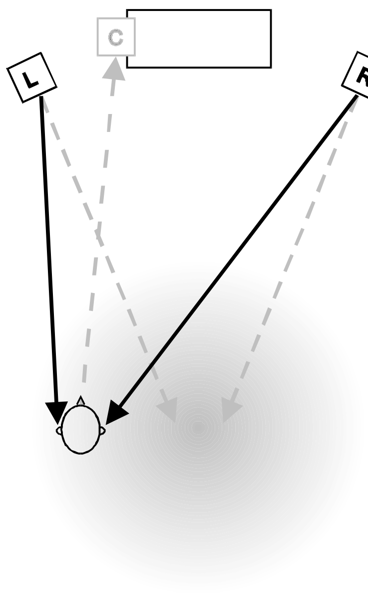


**Figure 2-9 Listener Shifted to Side**

This produces an eye/ear conflict, since the visual and audio images don't match. The addition of a Center speaker ensures that the center information, such as dialog, stays locked on the screen no matter where the listener is seated, as shown in *Figure 2-10*.

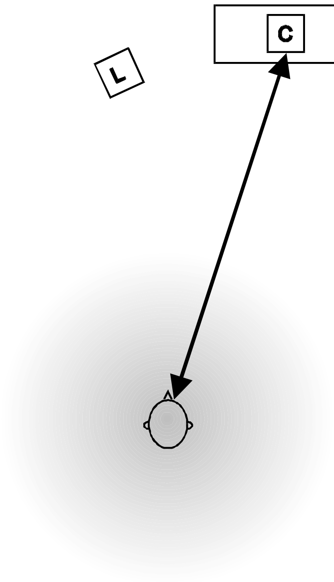


**Figure 2-10 Defined Image**

Also, since most engineers are used to mixing with a phantom center, it is easy not to realize how much mono or center information a typical mix contains. When the center speaker is added, it reproduces all mono information. A mix narrower than most people are accustomed to results. This further supports the need to have a Center speaker in the studio to hear what will result in homes with a Center speaker.

Place the Center speaker in the same horizontal plane as the Left and Right speakers whenever possible, *as shown in Figure 2-11*. In near-field applications, this is usually a simple task.

**Figure 2-11 Front Speakers in the Same Horizontal Plane**

When soffit mounted speakers are used, conflicts with video monitors may cause difficulty. If it is not possible to put the speakers in the same horizontal plane, place the Center speaker either above or below the video screen, as in *Figure 2-12* or *Figure 2-13*.

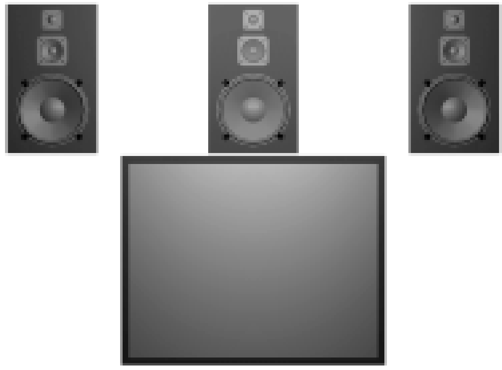

**Figure 2-12 Ideal Setup - All Speakers Above Screen**

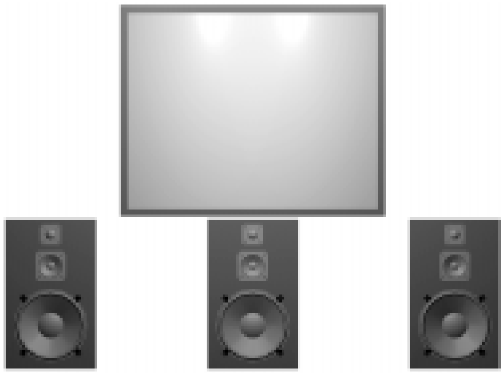

**Figure 2-13 Ideal Setup - All Speakers Below Screen**

The goal is to place the high frequency drivers (tweeters) in a straight line. This may require turning the Center speaker upside down or sideways, as shown in *Figure 2-14*. Make sure that the high frequency driver is oriented for the correct dispersion characteristics if you place it in any position other than its normal one.

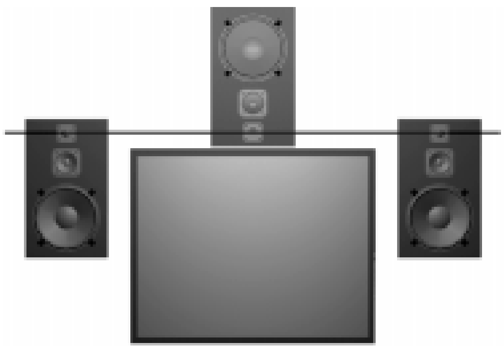

**Figure 2-14 Compromised Setup - High-Frequency Drivers In Line**

### 2.3.3 Smaller Center Channel Speakers

Many speaker product lines contain different sized models of the same design. The midranges and tweeters are normally exactly the same while the woofers differ in quantity and size. In cases where soffit space is limited, a smaller version of the main Left and Right speakers may be the only option for the Center channel. The Dolby SDU4 allows for the smaller Center speaker, with its reduced low frequency capabilities, by redirecting the Center channel low frequency information below 100 Hz to the Left and Right speakers. For further information on implementing this function, please see *Part 3.6.1 Bass Splitting Modification*.

### 2.3.4 Surround Channel Speakers

For normal Dolby Surround installations, small bookshelf speakers will suffice. However, you may wish to consider planning for the future. The 5.1-channel mixing format shown in Figure 2-15, is currently the format of choice for mixing motion picture.

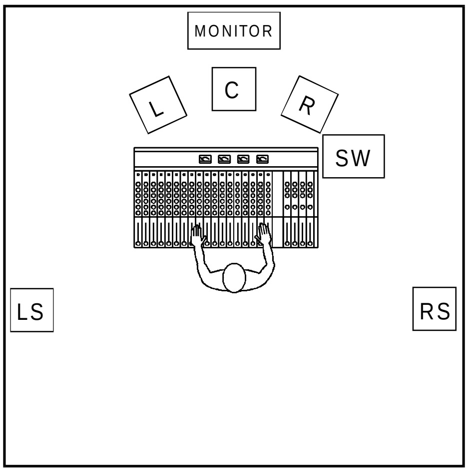


**Figure 2-15 5.1-Channel System Room Layout**

This format uses the standard three full range front channels, two full range Surround channels and one Low Frequency Effects (LFE) or *boom* channel. The LFE channel is band limited from 3 Hz to 120 Hz in the Dolby Digital format. A subwoofer, separate from any other front channel subwoofer, is normally connected to this channel with an appropriate amplifier. Since the Surround channels are independent (stereo) and full range, a little extra expense, wiring and thought may save headaches down the road.

To be 5.1-channel ready, use full range speakers in each location. If a smaller Center channel speaker is used, the same model may also be used for the Surround speakers. Use a separate power amplifier for each Surround speaker. Direct individual runs should be used for the audio wiring from each speaker to the amp rack or patch bay (for self-powered speakers).

### 2.3.5 Surround Speaker Location

For installations using one pair of speakers for the Surround channel, place the speakers on the side walls approximately two feet behind the engineer's seating position and at least two feet above the engineer's head. They should point to a spot two feet above the engineer's head, as in *Figure 2-16*. If four or more speakers are used, the same guidelines apply for each set of speakers. In any case, never point a Surround speaker directly at the listener or below their seating position.

**Figure 2-16  Vertical Location of Surround Speakers in Control Room**

### 2.3.6 Audio Consoles

The console's flexibility greatly affects surround mixing capability. While it is possible to create a Dolby Surround mix on a console with as little as a stereo bus and one auxiliary send, the ability to do complex mix moves is virtually nonexistent. A console with film-style panning allows the greatest flexibility for desired sound placement. Console automation also helps create complex mixes. The exact needs for a particular application will depend on the complexity of the mix. When deciding to purchase new equipment, it is a good idea to think about future needs, not just those of today.

### 2.3.7 Monitor Path

The normal audio path is from the console to the encoder to the recording device to the decoder to the speakers, as shown in *Figure 2-17*.

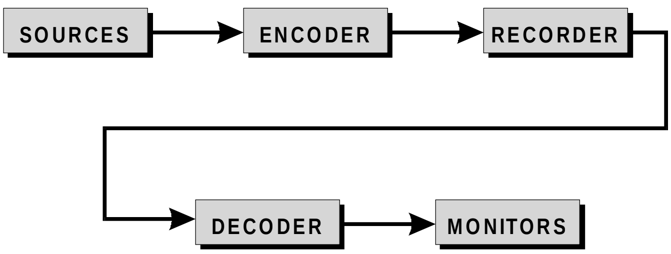


**Figure 2-17  Typical Signal Flow**

In most cases, this configuration disables console functions such as solo and source selection. To continue to use these functions, install the decoder in the monitoring path of the console, as shown in *Figure 2-18*. A few manufacturers have installed patch points in the proper place for this purpose. If you have this feature, follow the console manufacturer's instructions for installing the decoder. If you do not have this feature, some wiring or modifications may be required.

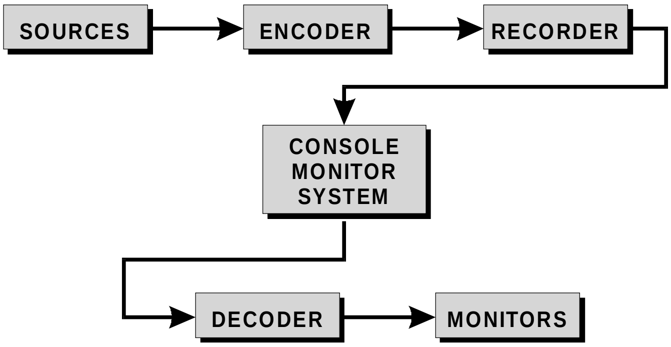


**Figure 2-18 Signal Flow through Console Monitor**

The simplest way to add the decoder is to connect it to the console's control room monitor outputs. If this is done, check three things. First, set the control room monitor level pot in a fixed position and leave it there. The decoder requires a calibrated reference level that changes if the control room level pot position does. Under this operation, the level control fader on the decoder (which can be remoted) becomes the new control room monitor level control. Second, the insert point for the decoder must be prior to any speaker switching circuitry for alternate speakers. Most consoles require modifications to add the insert points. Third, you may need to add switching circuitry for the extra speakers that are part of the surround monitoring system.

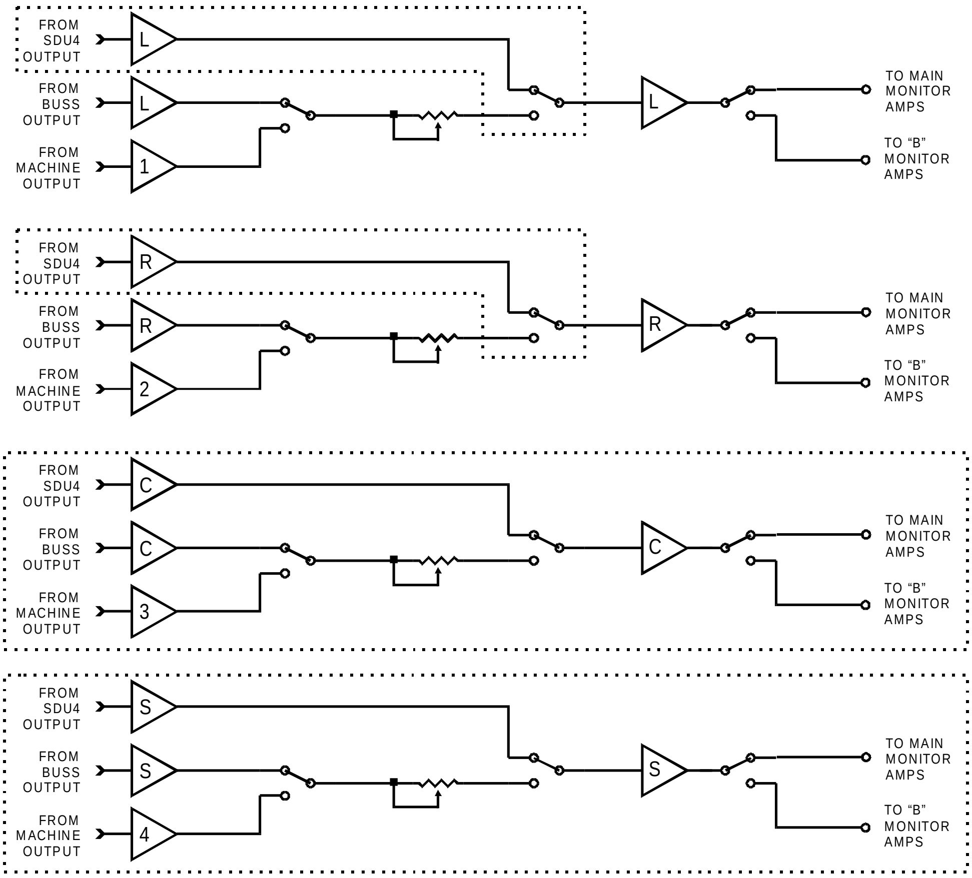


**Figure 2-19 Modified Monitor Section of 2-Track Console**

The other alternative is to only feed one set of speakers, usually near-field, when doing a Dolby Surround mix. In this case, the control room monitor output is fed to the decoder, which in turn feeds the speakers as was shown in *Figure 2-18*.

### 2.3.8 Speaker Sound Pressure Level

Speaker level is adjusted using the decoder's internal pink noise generator. The reference model for proper SPL is based on film practices. The SPMTE standards call for setting each channel so that pink noise at reference level reproduces at 85 dB SPL, C-weighted, slow. The following practices are based on this model.

For projects that involve preparing soundtracks for later film mixing and encoding with the Dolby Motion Picture Matrix system, set the levels at 85.

When mixing music, set the 0 dB reference levels at the same SPL in each channel. Some engineers like to mix more loudly than others, so as long as all channels are calibrated at the same level, the overall volume setting is not crucial.

When mixing in remote trucks and other very small mixing rooms where the surround speakers are in close proximity to the listener (typically 5 feet/1.5 meters or less), experience has proven that lowering the level of the surround speakers by 2 dB better represents the properly aligned typical home theater environment. This level change is only to be used in mixing environments and is not intended for general listening rooms.

When producing home video releases, experience has proven that lowering the overall mixing room level to as low as 79 dB results in a mix with better dialog levels. When mixing at 85 dB, mixers hear low level dialog easily in quiet control rooms, but consumers lose some of the quieter passages due to the higher ambient noise levels in homes. Competition with other people, appliances, and other sources of noise in the house tend to mask the low level dialog, making it unintelligible. At 79 dB mixing levels, the low level dialog is mixed louder so it can be heard in the mixing room. This results in a more consistent dialog level.

### 2.3.9 SPL Meters

A sound pressure meter is used to properly calibrate speaker levels. The most readily available units in the US are from Radio Shack, as shown in *Figure 2-20*. These units are also very inexpensive. Because the concern for level relative to each channel is usually greater than that for absolute level, the accuracy of this meter is sufficient. For greater accuracy, there are more expensive meters. It is recommended that an inexpensive meter be left in the control room for quick calibration checks.

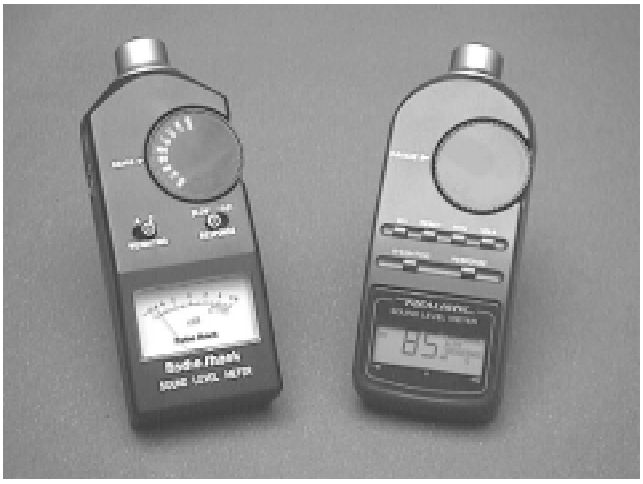

**Figure 2-20 Radio Shack Analog and Digital SPL Meters**

### 2.3.10 Phase Scope

A phase scope can assist in mixing. When the display is rotated 45 ° counterclockwise from the traditional display, as is available on the Tektronix 760 audio phase scope, the mixer sees graphically what is heard in the Dolby Surround sound field, as shown in *Figure 2-21*.


**Figure 2-21  Tektronix 760 Phase Scope Display of Center Channel Information**

The Left, Center and Right channels will appear across the top and the Surround channel will appear at the sides. Information that is in all channels will appear somewhat circular as in *Figure 2-22*. Individual channel information appears on the appropriate vectors.

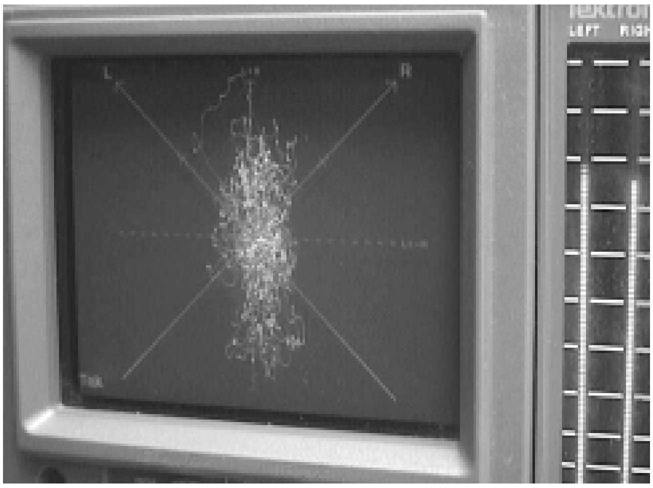


**Figure 2-22  Tektronix 760 Phase Scope - Typical Multichannel Information**

# Chapter 3
# System Installation

## 3.1 Signal Routing Audio Connections

### 3.1.1 Inputs and Outputs

The audio inputs and outputs of both the SEU4 and SDU4 use electronic floating balanced circuits and do not use ground as reference. They should be wired as you would an item with input/output transformers. In particular both signal pins (2 and 3) of the XLRs must *always* be connected. As the units are fully floating, either pin of an input or an output can be grounded, and the distinction between "hot" and "cold" is arbitrary, provided the same convention is used on all inputs and outputs. Current IEC wiring convention calls for XLR pin 2 to be "high/hot" and pin 3 "low/cold"; we suggest following this recommendation. *Figure 3-1* shows the pin arrangement; the unlabeled ring next to the pins is the shell connection.

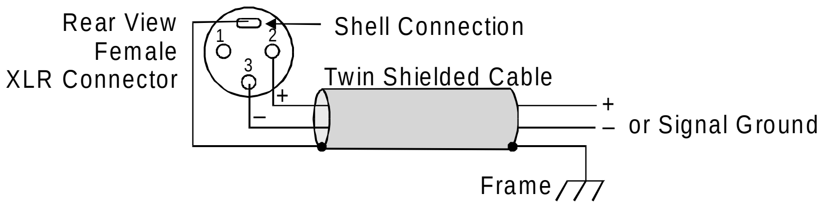


**Figure 3-1 XLR Connector Pins**

The input and output level potentiometers accommodate a range of nominal levels from –8 to +8 dBu.

### 3.1.2 Wiring for Maximum Immunity to Interference

The European Union now has mandatory standards for EMC immunity as well as emissions. Other countries are moving in the same direction. Past wiring practices and equipment designs, in the USA, Europe and elsewhere, have led to unnecessary susceptibility to RF interference and to hum. In particular, the connection of cable shields to internal signal grounds has resulted in RF and ground currents sharing paths with audio. In order to reduce hum, it has sometimes been necessary to disconnect the shield at one end so as to prevent 50/60 Hz ground currents from entering the equipment. This reduces hum but greatly increases susceptibility to RF.

Balanced floating circuits at one end at least of each interconnection and connection of cable shields to the *chassis* at *both* ends can largely eliminate these problems. Any 50/60 Hz or RF ground currents then flow in a separate path from audio currents. It is not necessary, nor even desirable, that the internal audio grounds of the two interconnected items be connected via the audio cables; they will ultimately be connected via the power safety grounds, which for safety reasons must never be removed, and may sometimes be at slightly different potentials.

Both the SEU4 and SDU4 have suitable floating input and output circuits, and may therefore be wired to minimize both RF and 50/60 Hz interference simultaneously, even if the other end of an interconnection is unbalanced.

We strongly recommend that all audio interconnections employ twin-core shielded cable. At the SEU4/SDU4 end, irrespective of the nature of the equipment at the other end, connect the inner wires to pins 2 and 3, and the shield to the shell of the XLR, not to pin 1. Leave pin 1 open-circuit. (Recent equipment has pin 1 connected to chassis, though usually with a higher inductance path than that via the shell, but the earlier versions of the SEU4/SDU4, like a lot of other audio gear, has pin 1 connected to an internal signal ground, providing a potential path for interference into the internal circuitry.)

If the equipment at the other end has balanced inputs or outputs, use the same convention there (inner wires to pins 2 and 3; shield to connector shell).

If the equipment at the other end is unbalanced, connect one of the inner wires to the “hot” or go side, the other to the “cold” or return, and the shield to the chassis. Sometimes, the chassis and the audio return are the same (e.g. RCA phono socket or other coaxial connector screwed to the panel), in which case one side of the audio and the shield go to the same place (at this end *only*). If in doubt, connect the shield by as short a path as possible to the chassis. The object is to ensure that the chassis of the items of equipment are interconnected via the shield, but that no audio currents flow in that shield.

The wiring is summarized as follows:

SEU4/SDU4 end:

| XLR connector | Cable |
|---|---|
| shell | Shield |
| pin 1 | Open |
| pin 2 | signal + |
| pin 3 | signal – |

Other end balanced:

| XLR/jack | Cable |
|---|---|
| shell/sleeve | Shield |
| pin 1/NA | Open |
| pin 2/tip | signal + |
| pin 3/ring | signal – |

Other end unbalanced:

| Coaxial connector | Cable |
|---|---|
| chassis, or shell if same as chassis | Shield |
| signal pin | Signal + |
| shell | Signal – |

**Note:**

Some equipment (not from Dolby Laboratories!) has balanced *non-floating* outputs on XLR connectors. Although they are described as balanced, they consist of two independent unbalanced outputs with respect to a signal ground, bearing equal audio signals of opposite polarity (resembling a center-tapped output transformer with the tap connected to signal ground). Such equipment must have pin 1 connected to signal ground, because if such an output is to feed an unbalanced input, it is necessary to use just one of the “unbalanced” feeds, and thus one of the two output pins as the “hot” and pin 1 as the “cold” (leaving the other audio pin open). Signal level is then reduced by 6 dB. Such an output can be treated as balanced only when it is connected to a balanced input (such as an item of equipment from Dolby Laboratories).

## 3.2 Signal Flow Options - Encoder

### 3.2.1 Basic Recording Setup with Auxiliary Bus Surround Feed

The most basic setup uses the encoder with a stereo output from the console feeding the left and right inputs and auxiliary buses feeding the center and surround inputs. The encoder then feeds to the recorder input. The recorder output feeds the decoder, which in turn feeds the amps and speakers. While this is the simplest way to encode, it is also the most limiting in terms of panning effects. This arrangement works best with live broadcasts and simple music mixes.

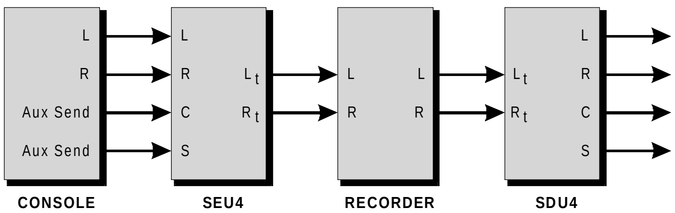


**Figure 3-2 Signal Routing - Stereo Bus and Auxiliary Sends**

**Troubleshooting Tip:**

Many consoles currently in production do not maintain consistent polarity on their outputs. (Many industry professionals refer to this polarity inversion as phase. In addition, many console manufacturers include phase buttons for each input channel, which allow the polarity of the signal to follow the wiring connections or, by enabling the switch, reverse or invert the connections via the switch contacts and therefore invert the polarity of the signal.) The auxiliary outputs of the console may not be in phase with the main stereo outputs of the console. When connecting reverb units, delays and other effects processors, absolute phase of these signals may not be a concern since it will not be maintained precisly after the effect is added. Unfortunately, this is a problem for Dolby Surround encoding. To check the signal polarity to the encoder, apply a 1 kHz signal to the left and right encoder inputs. While observing the encoder outputs, add the same 1 kHz signal to the Surround channel. The outputs should both increase in level as the surround input level increases. If one channel goes up and the other channel goes down, the polarity of the Surround channel input to the encoder is reversed. If you vary the frequency of the signal, you will get varied results between the two channels, depending on the frequency of the tone. To correct this problem, reverse the polarity of the input to the encoder Surround channel input by swapping the connections to pins two and three.

### 3.2.2 Basic Recording Setup with Film Panning Console

The most versatile console setup uses a console with film-style (LCRS) panning. These consoles can pan from left to center to right and from front to back. This pan pot system can place sounds quickly and easily. These consoles will have left, center, right and surround outputs for connection to the encoder. The output of the encoder follows a flow similar to the above example.

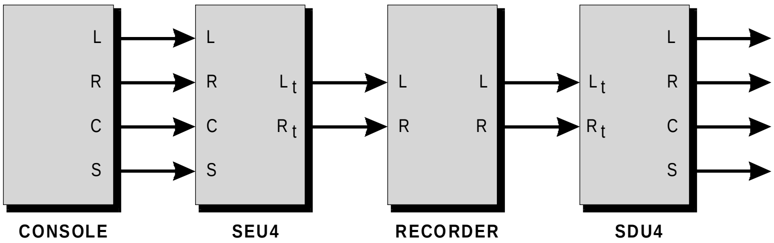


**Figure 3-3 Signal Routing - Film-Style Panning**

### 3.2.3 Basic Recording Setup with 2 Stereo Bus Output

Some consoles have multiple stereo buses, as is common with broadcast consoles. In this case, one stereo output can be used for left/right panning and a second used for center/surround panning. Although not as flexible as a film style panning setup, this configuration will serve the needs of most applications with few limitations.

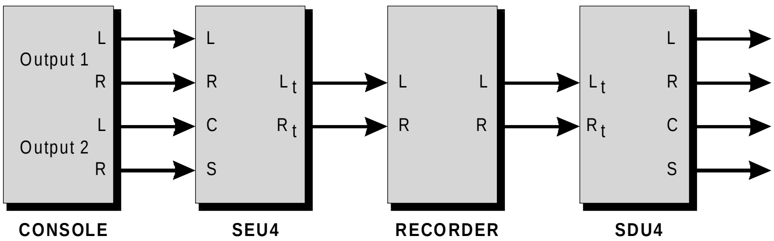


**Figure 3-4 Signal Routing - Dual Stereo Bus**

## 3.3 Signal Flow Options - Decoder

### 3.3.1 Recording Setup with Monitor Section of Console

All of the above connections involve feeds to the encoder. They assume a signal path from the encoder output to the recording device. The recording device then feeds the decoder that in turn feeds the amps and speakers.

For installations where the console contains a monitor section, all monitoring functions such as solo, dim and source selection will be lost. To restore the monitor operations in the console, connect the units as shown in *Figure 3-5*.

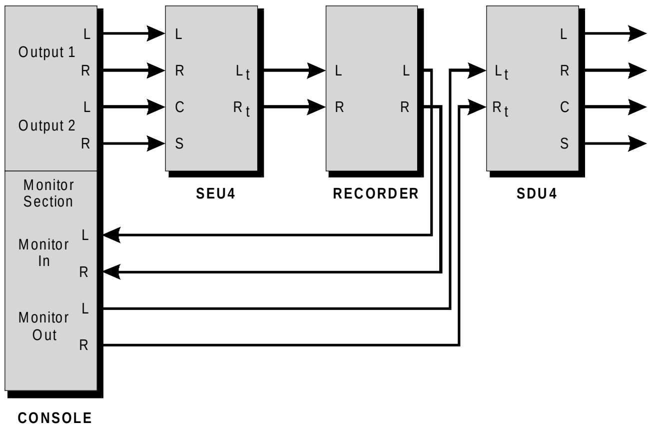


**Figure 3-5 Signal Routing - Console with Monitor Section**

The only caution here is that the console monitor level must be set and the decoder level calibrated. Once this is done, do not move the console monitor level control. Use the level control on the SDU4 or install a remote level pot. It may or may not be possible to physically insert the SDU4 remote pot in place of the current monitor pot. See *Part 2.3.7* for further information.

### 3.3.2 Recording Setup with Surround-Ready Monitor Section of Console

Some newer consoles are equipped with a multichannel volume control and monitor loop insert points for inserting the SDU4. In this case, installation is very simple and the monitor level control is post the insert point. In this case, the console pot can be used to control overall gain, as shown in *Figure 3-6*.

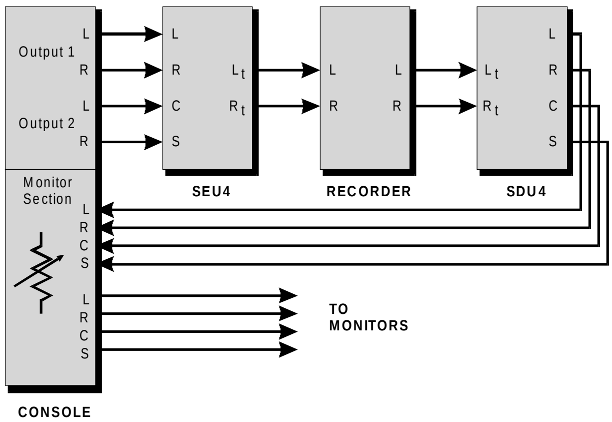


**Figure 3-6 Signal Routing - Monitor Section with 4 Channel Insert Points**

### 3.3.3 Live Broadcast Setup

Setups for live broadcast are the same as studio setups, except that generally, the signal is fed to the station instead of to a recording device. This is not to say that the event could not be recorded locally at the same time. In these cases, any of the above encoder wiring schemes are possible. The decoder is normally fed from the distribution amp system used to feed the transmission path. See *Figure 3-7*.

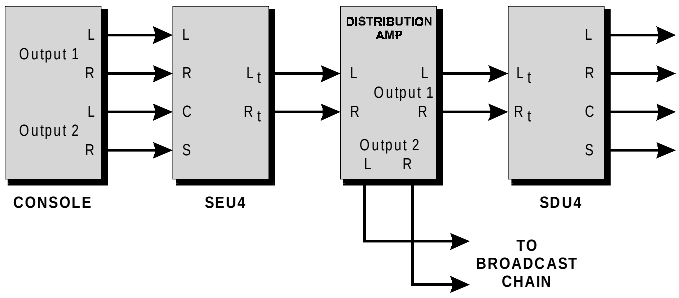


**Figure 3-7 Signal Routing - Live Broadcast**

### 3.3.4 Live Broadcast Setup with Fail-Safe

Because of the addition of patch points to insert the Dolby Surround encoder in the final outputs of the console for typical live broadcast applications, many mixers have adopted the use of a fail-safe connection. This method requires a larger console, usually one with multi-track outputs, to accommodate both the extra outputs and inputs required to use this method. See *Figure 3-8*.

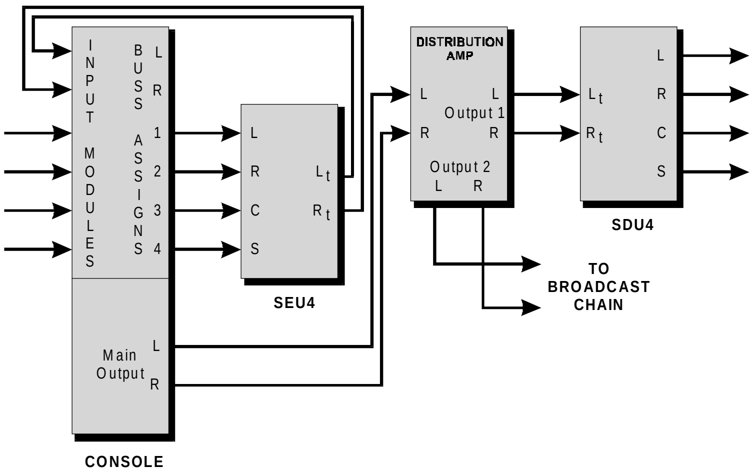


**Figure 3-8 Signal Routing - Live Broadcast with Fail Safe**

The premise is that if a patch or the encoder should fail while on the air, the modules assigned to feed the encoder can easily be reassigned to the stereo bus of the console and audio can be re-established very quickly. Although the signal will no longer be Dolby Surround encoded, at least audio will be on the air. Once the program goes to a commercial or break, repairs can begin. Either connection system will work, but one has redundancy for the unexpected, while the other does not. While Dolby products rarely fail, patch bays and patch cords are another matter.

### 3.3.5 Monitoring Music Premixes for Film

4-2-4 monitoring is used primarily in the production of music soundtracks for film work. Because these tracks are normally sent to the final mix as separate elements rather than a complete mix, these signals are not actually encoded when preparing the elements for delivery to the mixing facility. In order to ensure that there are not any surround compatibility problems with the elements, they are mixed through the console and fed through an encoder and decoder to the speakers in the room. In this case, the output of the encoder feeds directly to the decoder. Both units are in the monitoring chain, not the recording chain. Such monitoring can also be used when tracking a music session that will be mixed later in Dolby Surround.

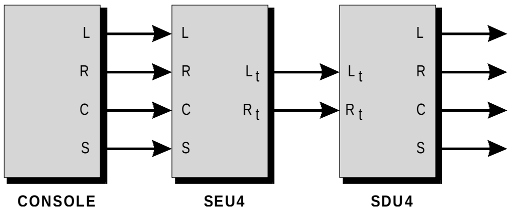


**Figure 3-9 Signal Routing - 4-2-4 Monitoring**

## 3.4 Dolby Model SEU4 Setup

Normal operation of the SEU4 requires no modifications to the unit as it comes from the factory. There are two modes which should be checked if problems are encountered.

On the older units ("gold" finish), access the switches referred to below by removing the front cover plate (using the thumbscrews) and withdrawing the plug-in modules. On the more recent units ("silver" finish) remove the front extrusion first. However, it is likely that these switches are in their default positions.

### 3.4.1 Surround Active LED

The Surround Active LED on the front should be lit as in *Figure 3-10.* If it is not, a jumper inside the unit has likely been changed (there is no reason to do this under normal operation) or a jumper wire (or closed switch) between pins 5 and 15 of the DB15 connector on the back of the unit is disabling the function. Open this switch or remove the jumper to restore operation.

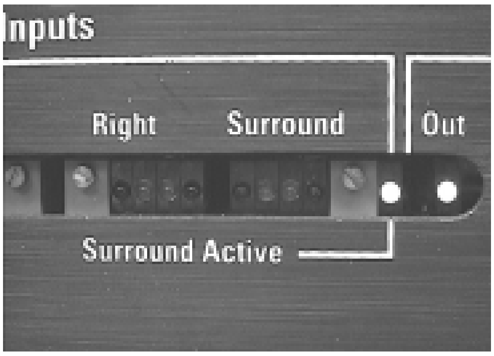


**Figure 3-10 SEU4 Front Panel with Surround Active LED On**

**Note:**

If your unit is modified for game mode *(Section 7.5)* the surround active light indicates normal operation. If the LED is not illuminated, the unit is in game mode.

### 3.4.2 External Processing Loop (EPL) Loop Switch

The SEU4 can be used with an external processor inserted in a loop before the final output stages of the unit, *Figure 3-11*. In normal operation, this loop is unused. The processor is placed after the encoder outputs and before the next device in line such as the stereo master fader on the console. The EPL loop output is a separate output with its own level controls. External equipment can be inserted into the loop either by moving the internal switch labeled EPL to the “IN” position, or more conveniently (leaving the switch “OUT”) by linking pins 6 and 15 on the remote control connector J507 on the rear of the unit.

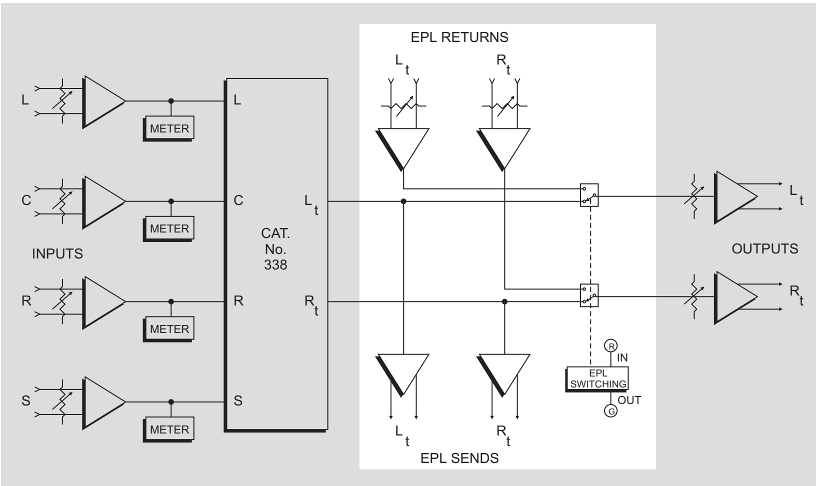


**Figure 3-11 SEU4 with EPL Highlighted**

**Troubleshooting Tip:**

If you are feeding signals to the unit, but there are no output signals from the main output connectors for both channels, chances are that the loop is switched in and you don't have any EPL connections in place to complete the signal path. Move the EPL switch to the **out** position to restore output to the main output connectors.

## 3.5 Dolby Model SDU4 Setup

### 3.5.1 Internal Switches - CAT 344 information

Several switches and jumpers on the right card inside the SDU4 may need to be checked for proper settings. These switches cover the wake up mode, Center speaker status, local or remote fader and external processor loop. The factory configures them with Dolby Surround as the wake up mode, Center speaker on, remote fader disabled and no external processor loop. Remove the card to make any changes necessary. On the more recent units (“silver” finish) it is necessary to remove the front extrusion to gain access to these switches.

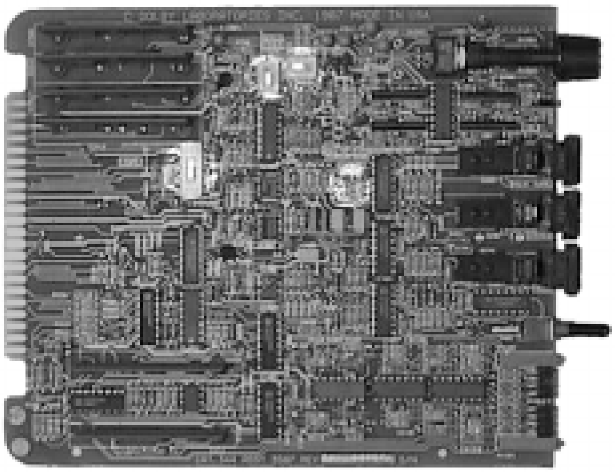

**Figure 3-12 CAT 344 Card Switches and Jumpers**

### 3.5.2 Center Speaker Switch

When using the SDU4 with a Center speaker, the switch should be in the **yes** position, *Figure 3-13*. This is the recommended configuration. In this configuration, there will be Dolby Surround-decoded Center channel audio information from the Center speaker when in the Dolby Surround mode, no audio in the Center speaker when in the Stereo mode (conventional two-channel stereo from the Left and Right speakers), and a Mono summation of the audio in the Center speaker when in the Mono mode.

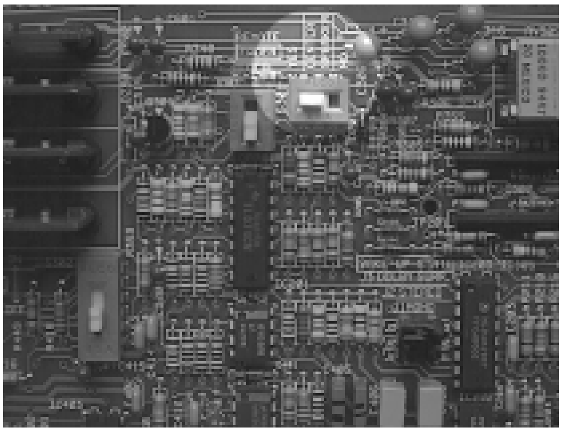

**Figure 3-13 Center Speaker Switch Detail**

With the switch in the **no** position the Dolby Surround decoded Center channel audio information will be added equally to the signals for the Left and Right channels and will be heard from the Left and speakers as a phantom image for the Dolby Surround mode, no audio in the Center speaker when in the Stereo mode (conventional two channel stereo from the Left and Right speakers), and a Mono summation of the audio in the Left and Right speakers for Mono mode.

**Troubleshooting Tip:**

If a Center speaker is not in use and left and right information can be heard, but center information can't be when in either the Dolby Surround mode or Mono mode, the Center speaker switch is probably set to the yes position. This causes all of the center information to be fed to the center output of the decoder (which, in this case, is an open unconnected output). To correct this, remove the card and move the Center speaker switch to the no position or add a Center speaker and amplifier.

### 3.5.3 Wake-up State

The unit is set at the factory to wake up in the Dolby Surround mode when power is applied. If you want the unit to wakeup in the stereo or mono mode, remove the card and move the jumper to the appropriate position, as shown in *Figure 3-14*.

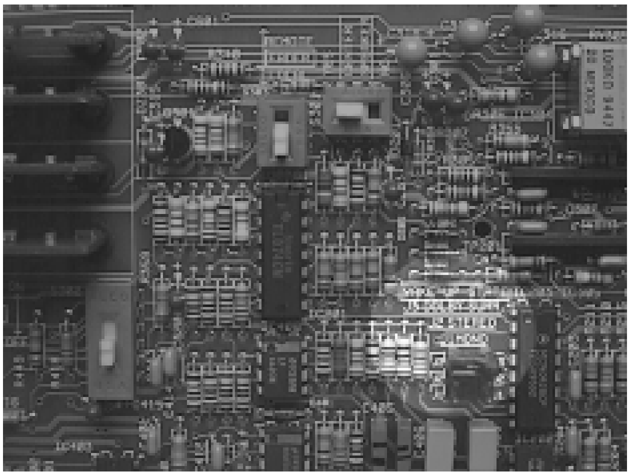

**Figure 3-14 Wake-Up State Jumper Detail**

### 3.5.4 Local/Remote Fader

The master volume control on the front of the unit controls the decoder's four outputs. Alternately, you can add a fader in a remote location by connecting a 100kΩ pot to the DB25 connector on the back, as shown in *Figure 3-15*. To activate the remote fader, move the remote fader switch, as shown in *Figure 3-16*, to the **remote** position.

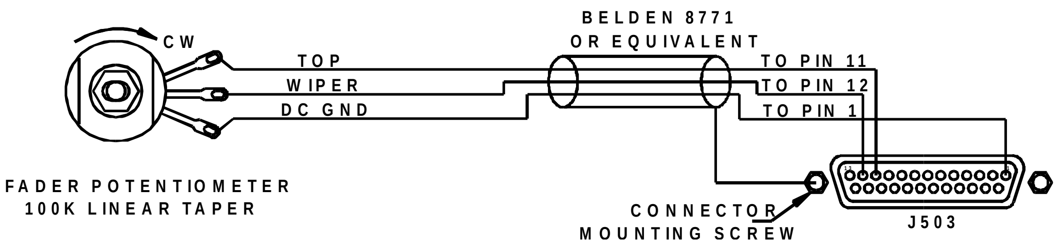


**Figure 3-15 Remote Fader and Connector**

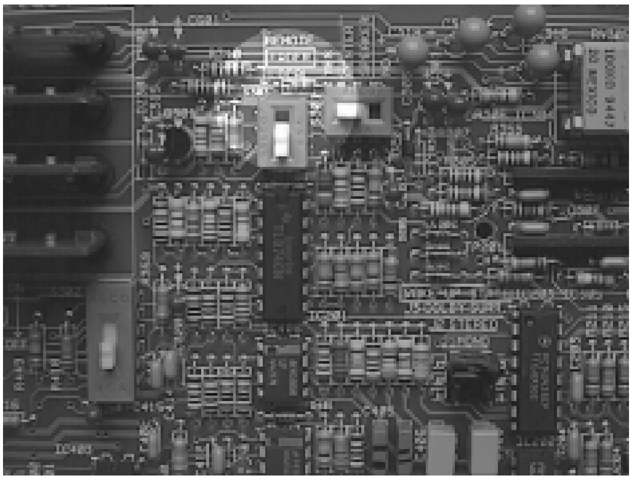

**Figure 3-16  Remote Fader Switch Detail**

**Troubleshooting Tip:**

If the volume control on the unit does not seem control the monitor level, check the remote fader switch. The unit will output audio at full volume with the switch in the remote position and no pot attached.

### 3.5.5 EPL Switch

As with the encoder, the decoder contains an EPL switch as shown in *Figure 3-17*. The older units ("gold" finish) had an external processor loop selected by an EPL switch on the Cat.No.344 module. See *Figure 3-17* and Table 3-1.

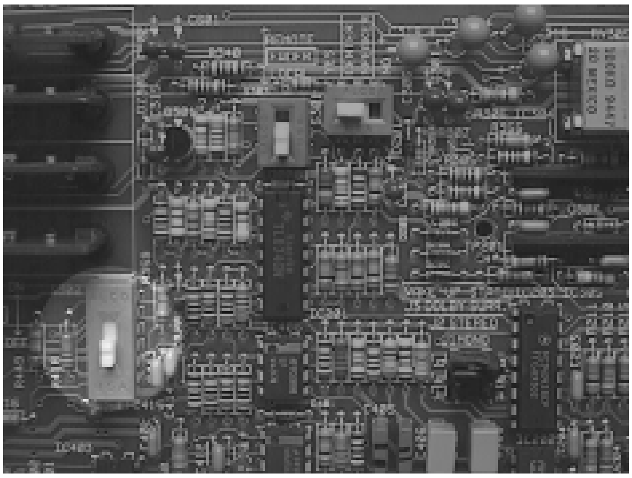

**Figure 3-17  EPL Switch Detail**

The connections are made via the DB25 connector on the back. The connections are as follows:

**Table 3-1  EPL Connections**

| Pin | Signal |
|---|---|
| **1** | Ground |
| **2** | Loop Send Surround |
| **3** | Loop Send Right |
| **4** | Loop Send Center |
| **5** | Loop Send Left |
| **14** | Loop Return Surround |
| **15** | Loop Return Right |
| **16** | Loop Return Center |
| **17** | Loop Return Left |

Later units (“silver” finish) only have the sends, usable for level monitoring; the loop returns are omitted. The switch on the Cat.No.344 module must therefore always be left in the factory default **off** position.

**Troubleshooting Tip:**

If audio is being sent  to the unit and the master level control is set correctly, but no audio appears at the outputs, check the EPL switch. It is probably in the in position. Remove the card and move the switch to the out position.

## 3.6 Cat 150E Card Settings

### 3.6.1 Bass Splitting Modification

In most studio applications, all three front speakers will be identical. In some applications, however, a smaller version of the Left and Right speakers must be used for Center. If possible, the midrange and tweeters in the three speakers should be identical. If this is the case, the center woofer will be smaller. Because most bass information is normally found in both the Left and Right channels, panned center, the decoder will place these low frequencies in the Center speaker. The decoder contains a special feature that can take the Center channel low frequency information below 100 Hz and redistribute it to the left and right outputs where the larger woofers exist. To enable this feature, install a soldered jumper wire between pin T of the Left card (CAT 150E) and the unbanded end of the diode (D401) adjacent to pin T, as shown in *Figure 3-18* Do not get solder on the gold edge card pin except at the very end (away from the edge of the card).

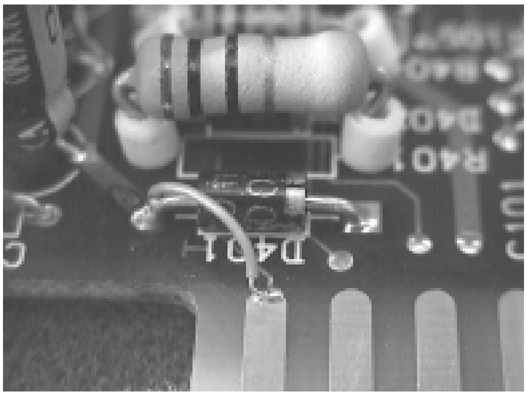

**Figure 3-18 Detail of Jumper Modification for Bass Splitting**

### 3.6.2 Time Delay Calculations

In addition to the switches mentioned above, a rotary switch sets the time delay to the Surround channel, as shown in *Figure 3-19*. On older units, the switch is behind the removable front panel cover. On newer units, the switch is visible through the round hole in the front panel.

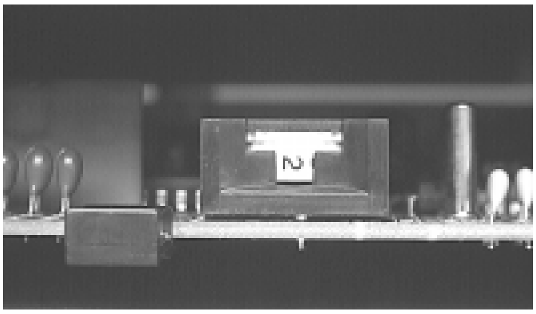

**Figure 3-19 Delay Switch Detail**

To calculate the proper delay, measure the distance in feet from the seating position to the front speakers, and subtract the distance from the seating position to the nearest Surround speaker. Add 10 to the result. This is the proper time delay in milliseconds. The minimum delay available is 20 ms. If your answer is less that 20, use the 20 ms setting. Always round up to the next available delay time. For example, 18 becomes 20 ms and 22 becomes 30 ms. For most trucks and small studios, 20 ms will be the proper delay time.

In metric measurements, measure the distance in meters from the listening position. Subtract the distance in meters from the listening position to the nearest Surround speaker. Multiply this number by 3 and add 10 to that answer. This is the delay time.

The zero setting is 20 ms and each number upward adds 10 ms.

**Table 3-2 Delay Switch Settings**

| Setting | Delay |
|---|---|
| 0 | 20 ms |
| 1 | 30 ms |
| 2 | 40 ms |
| 3 | 50 ms |
| 4 | 60 ms |
| 5 | 70 ms |
| 6 | 80 ms |
| 7 | 90 ms |
| 8 | 100 ms |
| 9 | 110 ms |
| 10 | 120 ms |
| 11 | 130 ms |
| 12 | 140 ms |
| 13 | 150 ms |

# Chapter 4
# System Set-Up

## 4.1 Encoder Alignment

Correct electronic alignment is a must for proper surround mixing. Perform the following steps on initial installation of the equipment and verify system integrity from time-to-time.

1. If it isn't already, connect the unit to an audio path. See *Section 3.2* for further information. If using the effects processor loop, switch it out for the following alignments. You will be instructed when to switch it back in. (To disable the EPL, move the slide switch located front center of the right-hand board, Cat.No.385, or open the link between pins 6 and 15 on the remote control connector.)

2. Apply a 1 kHz tone at console reference level (+4 dBr, 0 VU, etc.) to the Left channel input.

3. Adjust the Left channel trim control until both green LED's on the SEU4 are illuminated. The resolution from left green LED to right green LED is approximately 1/4 dB. In some cases it may be difficult to keep both green LED's lit. If so, adjust until you are at the crossover point from one green LED to the other.

4. Adjust the Lt output trim pot to reflect console reference on the metering employed, console or recorder. If you are returning the signal through the console master fader, be sure the fader is set for unity gain. Once the master fader is set, do not change it for the remainder of the setup procedure.

5. Apply the 1 kHz reference tone to the Right channel input.

6. Adjust the Right channel input trim and Rt output trim as in steps 3 and 4 above.

7. Apply the 1 kHz reference tone to the Center channel input.

8. Adjust the Center channel input trim, as shown in *Figure 4-1*, to light both green LED's. Do not adjust the output trims for Lt or Rt. The left and right meters on the device being fed by the encoder, console, or recorder should both read approximately -3 dB and the signal should be in phase.

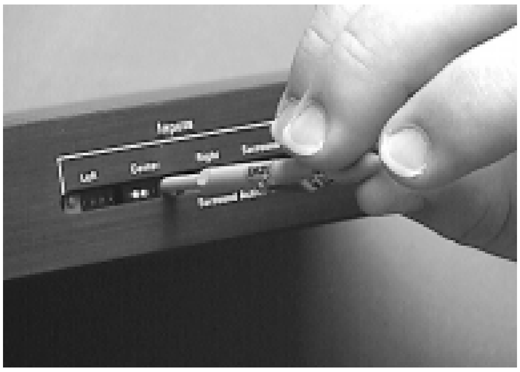

**Figure 4-1 Adjusting the Center Input Trim Control**

9. Apply the 1 kHz reference tone to the Surround channel input.
10. Adjust the Surround channel input trim to light both green LED's. Again, do not adjust the Lt or Rt output trims. The left and right meters on the device being fed by the encoder, console or recorder, should both read -3 dB and the signal should be 180° out of phase.
11. If the Effects Processor Loop (EPL) is not used, encoder alignment is complete, proceed to decoder alignment.
12. Switch the EPL in.

The EPL contains send and return levels and is used to interface a piece of signal processing gear after the encoding to Lt/Rt, but before the final output of the SEU4 encoder. These trims are usually set for unity gain at the factory. Should you desire to change them, apply the 1 kHz reference signal to the left and right inputs and adjust left and right EPL sends for the proper level at the signal processing device input. Then adjust the left and right returns to produce the proper level at the SEU4 output. To enable the EPL, move the slide switch located front center of the right-hand board, Cat.No.385, or link pins 6 and 15 of the remote control connector.

## 4.2 Decoder Alignment

The decoder has two parts to align: the input levels and the output levels.

### 4.2.1 Input Levels

1. Feed a 1 kHz tone to the left and right inputs of the decoder. (This should be fed to the encoder, which in turn should feed the decoder as well as the rest of the signal chain.)

2. Adjust the left and right inputs, as shown in *Figure 4-2*, to light both green LED's. The inputs are now aligned.

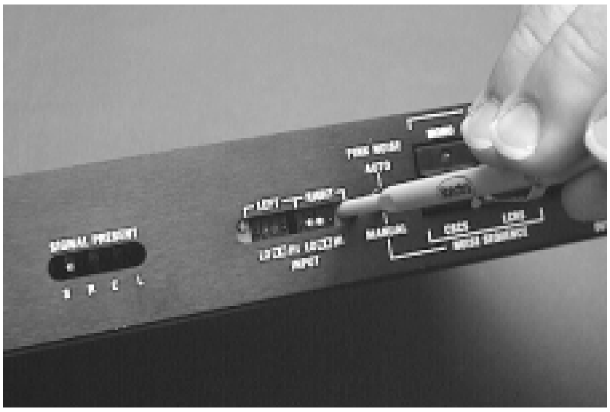

**Figure 4-2 Adjusting the Right Input Trim Control**

### 4.2.2 Output Levels

To adjust the outputs to the speakers, set the volume control to the reference position, about 2 o'clock on the scale, and use the built-in noise generator to send pink noise to each speaker. Adjust each speaker level (usually done at the amplifier) to obtain 85 dB SPL, C-weighted, slow. If you do not have a Center speaker, ignore the setting for the Center speaker and adjust only Left, Right and Surround. The level for Center should then be correct. Slight level variations may be caused by the acoustical environment. See *Part 2.3.8* for further details on modifying the 85 dB calibration level.

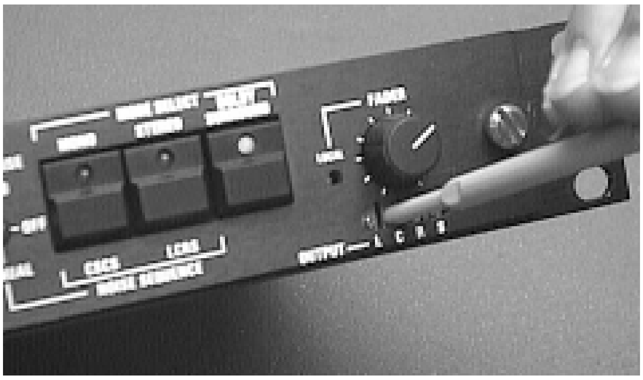

**Figure 4-3 Adjusting the Center Output Trim Control**

## 4.3 Room EQ

Mixing room designs always include some form of room equalization. The most common way to achieve it is to use multi-band equalizers. In addition, wall and ceiling treatments may be required. The use of near field monitors has become popular recently because the room environment does not adversely affect the near field monitors. .

### 4.3.1 ANSI/SMPTE 202M X-Curve

When in large mixing rooms, defined as greater than 5300 cubic feet or 150 cubic meters, use the X-curve as defined by the ANSI/SMPTE 202M standard. This curve is flat from 63 Hz to 2 kHz and then falls off at 3 dB per octave from above 2 kHz. See *Figure 4-4*. There is also a 3 dB roll off on the bottom end, with 50 Hz being down 1 dB and 40 Hz down 2 dB.

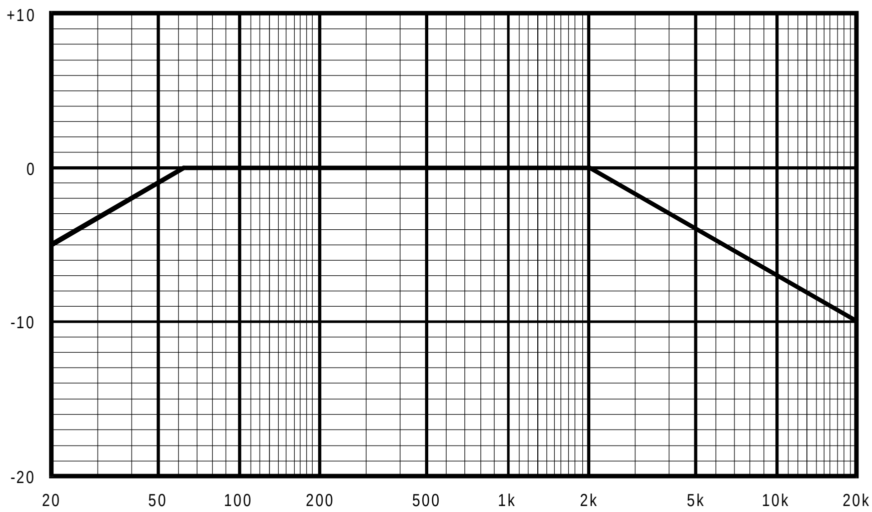


**Figure 4-4 Standard X-Curve**

### 4.3.2 ANSI/SMPTE 222M Modified X-Curve

For small rooms, defined as less than 5300 cubic feet or 150 cubic meters, ANSI/SMPTE 222M calls for a modification of the X-curve with flat response to 2 kHz and then a 1.5 dB per octave roll off above 2 kHz. See *Figure 4-5*. This curve is useful when mixing in a small room and be playing back in a large room. Some people prefer to use the low end roll off as defined by the standard; some prefer to leave the low end flat. It is generally left flat. Another variation on the curve is to begin the high end roll off at 4 kHz instead of 2 kHz and roll off 3 dB per octave instead of 1.5 dB per octave.

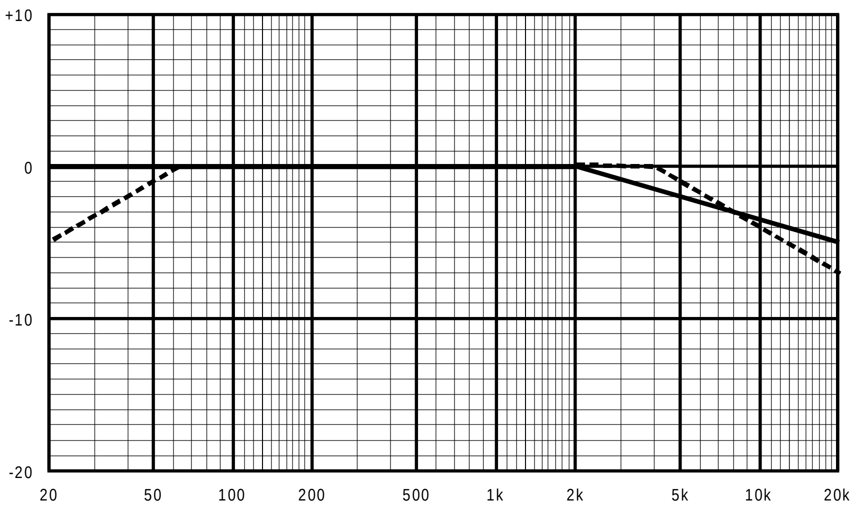


**Figure 4-5 Modified X-Curves**

### 4.3.3 Recording Studios - Music Mixing

There are perhaps more arguments over what equalization should be used in a recording studio than any other place. Most studio designers have an opinion about what the curves should be and how they relate to what is ultimately heard by the end user. Find out what initial curve your room designer used to set up your facility for music mixing. Most designers tend to roll off a little of the top end frequency response. Some prefer to start at 8 kHz and roll off such that 16 kHz is down 3 to 4 dB. Others believe that the system should be flat to 12 kHz and roll off a few dB per octave from there.

No matter what convention you choose, equalize the Left, Center, and Right speakers to match the curve. If you are using a small Center speaker with the bass splitting modification; see *Chapter 3*. In this case, the low end of the Center speaker will not be equalized below 100 Hz, as this information is carried by the Left and Right speakers.

### 4.3.4 Near-Field Monitors

Because near-field monitors do not generally interact with the room at the listening position, no room equalization is usually required or used. Several self-powered near-field monitors include equalization adjustments. Follow the guidelines above for setting the equalization adjustments. The flat position usually refers to a measurement taken at 1 meter from the speaker that reproduces the designers desired response.

# Chapter 5
# Mixing Techniques

## 5.1 Announcers and Dialog

Traditionally, dialog is placed only in the Center speaker to tie the on-screen sounds to the picture. When a Center speaker is used, all center-panned dialogue appears to come from the screen regardless of the listener's position. If the dialogue comes from the Left or Right speakers, the stereo image differs depending on the listener's position. This is highly undesirable. It does not bar voices from the other channels, but generally only effects or incidental voices should be in any channel other than center.

## 5.2 Interior Effects

Interior sound effects come from all four channels and appear to surround the listener. Wind noise, crowds, and other general ambient sounds are included within the mix to give a sense of realism. Effects and ambient sounds will normally appear in the Left, Right, and Surround channels. It is common to use Stereo ambiance that is panned left and Surround for the Left channel of the source and Right and Surround for the Right channel of the source. The resulting sound surrounds the listener, yet still has a front stereo image. The amount of Surround channel signal added determines how far back the listener is in relation to the front sounds. More surround level produces an image that sounds further back in the room.

Sometimes only a mono element is available when a surround effect is desired. In this case, put the signal in both the Center and Surround channels. This is commonly known as a 2-4 punch. With equal application to both Center and Surround, the sound appears to come from all four channels. Stereo reverb can also give a mono sound element a slightly wider image. Simply apply the reverb effect to the Left and Right channels while applying the original dry signal to the Center and Surround channels.

Another effective technique is to and assign the signal to the Left channel and add about 8 milliseconds delay of the same signal to the Right channel. This may or may not produce acceptable results, depending on the program material.

## 5.3 Positioning of the Stereo Image

The Center speaker in a Dolby Surround system produces stereo imaging that is slightly different than that of a two speaker stereo system. Most music engineers find this distracting at first, but adjust quickly. Those who mix motion picture sound feel comfortable, as do those music engineers who own a home theater system. The most noticeable difference in the stereo image is that the perceived image tends to be narrower when a Center speaker is used. Because most music mixes contain significant amounts of Center channel information, we are used to hearing a phantom image produced by the Left and Right speakers. Since all of this information is now directed to a single point source, the Center speaker, we perceive it as all center. To correct this in the mix, make the image slightly wider than normal for a two-channel stereo mix.

Do not eliminate the Center speaker in the control room and use the phantom monitoring mode. While this may produce a more familiar sound in the control room, it does not satisfy home listeners a who have a complete home theater system. To achieve the correct listening result, mix with a Center speaker.

## 5.4 Panning Sounds

There are several ways to pan sounds in a mix. It is most effective to use film-style multichannel panners from left to center to right, and from front to back. This allows you to position a sound anywhere in the sound field with little effort.

If film style panners are not available, panning from left to right on one stereo bus and center to surround on a second stereo bus is also effective in sound placement, although this technique is not as easy to use, especially for moving effects.

On small, function limited consoles, use the stereo bus for the front channel panning and feed the Surround channel with an auxiliary bus. This is extremely limiting, but will work if necessary. For complicated panning moves, bring up the signal on more than one fader, and fade each to a different output, which allows you to use the faders as panners.

For example, if you assign one fader to the left/right pan and set the pan pot halfway between the left and center, assign another fader to the surround bus. This lets you pan from left center to surround by bringing up the fader feeding the Surround channel as the fader feeding the left/right buses is brought down. If the console is automated, these moves can be perfected individually and then repeated by the automation system. This method also works on larger consoles that require automated pans.

Assigning the signal to four faders, with each fader assigned to a different input of the encoder, can also create a circle pan.

## 5.5 Stacking Encoded Tracks

In the film industry, it is common to premix elements for the final mix. This can be done in the opening sequence for a series of shows or for a sound effect panning through the room.

The individual elements may be mixed as Dolby Surround encoded two-channel elements (Lt/Rt) and all of those elements may be mixed together in the final mix. Each element should then be assigned to the left and right inputs of the encoder.

## 5.6 Magic Surround

In certain cases, stereo microphone placement techniques and stereo electronic instruments cause a phenomenon known as *magic surround*. The decoder will decode some of the signal placed it in the Surround channel. This results in out of phase or inverted information in the stereo pair, which XY stereo microphone techniques typically produce. While this may sound pleasing by itself and no encoding seems necessary, this is an unpredictable process that should not be relied upon. Adding another element to the mix, such as a voiceover, could easily change the mix’s phase characteristics of the mix and alter the decoding process. It is best to put at least a little information from this signal source through the surround input of the encoder, ensuring the decoding of the real surround signals, not some random out of phase information.

In some cases, too much surround information is present, as with electronic keyboards, which achieve a stereo signal from a mono source via electronic processing. . If your favorite sound produces too much information in the Surround channel, simply pan the left a little towards center and the right an equal amount towards center. This cancels out some of the out of phase information and corrects the decoder. The panning required varies with the sound, but usually does not take much to produce a good result.

## 5.7 Decoder Mistracking and Steering Artifacts

When mixing, the decoder can only steer in one direction at a time, so you must plan the sound field carefully. Movie mixers have been doing this successfully for over 20 years, so do not be overwhelmed -it can be done quite easily.

Problems result when two very different and unrelated sounds are sent to two different channels at once. For example, crickets in the Surround channel and chickens in the front cause the sounds to bleed into the other channels and produce a dynamic image shift. This effect is distracting and undesirable.

It is also common for a music track to contain a prominent lead instrument in the Left or Right channel while an announcer comes from the Center channel. The instrument appears to move from its intended speaker towards the center when the announcer speaks and then return to the correct speaker when the announcer stops talking. The solution is to either pan the music element towards center or temporarily reduce its level. .

Often, when producing motion picture sound effects, the sound effects designer removes all ambient sounds briefly so that another may be heard more prominently. . For example, background sounds might contain a little traffic. A door slam may be next. At this, the traffic and night sounds are either very low level or disappear entirely. After the door slams the other sounds are already back in the mix. Because the door slam covered them, they aren’t missed by the audience.

## 5.8 Surround Pumping

Bad transmission paths frequently cause pumping of the Surround channel that is rarely heard in the mixing environment. Often a limiter is active in one channel of the transmission path but not the other, or a stereo limiter is not set up the same for both channels. The solution is to either remove the limiters or set them up identically and verify that they are linked together. This problem can be heard during the mix if a stereo limiter is being used excessively or if only part of the element is limited. It is impossible to discuss all possibilities, but always look for inconsistencies between the two stereo channels. The viewer usually recognizes the problem as Surround channel ambience pumping in response to the dialog. This is particularly noticeable during live sporting broadcasts when there is crowd noise in the Surround channel.

## 5.9 Proper Surround Level and Content

When is there enough surround content? This decision is usually left up to the taste of the producer and engineer mixing the project. As a guideline, the image should direct attention to the front of the sound field and it should be noticeable when the Surround channel is removed from the mix. Attention should not be drawn directly to the Surround channel when it is returned to the mix. Surround channel effects should complement, not distract from on-screen action.

## 5.10 Limiters, Delays, Reverb Units, Other Effects Processors

As with any mixing situation, signal processing devices are common in Dolby Surround mixes. Limiters and compressors cause few side effects if they are used before the encoder. Digital delays and sound field generators, reverbs, and so forth, may also be used. However, the tricks that generate the sound fields from these effects may not work as expected when Dolby Surround decoding is used. Since you are monitoring through a decoder, you can instantly hear what the sound field will actually sound like. If you find that your favorite reverb program has excessive surround content before anything is sent to the Surround channel, remove the stereo output from the device and pan it a little towards the center instead of hard left and right. Experiment to get the desired sound. The phase shifting of the effects unit competing with the phase encoding found in Dolby Surround causes this effect, which stereo keyboards also have.

## 5.11 Mono to Stereo Synthesizers

Mono to stereo synthesizers can create all sorts of havoc in a Dolby Surround mix. First, the Dolby Surround program *is* a stereo signal so there is no need for a synthesizer in the transmission path. Second, if you have ever listened to a mono show run through an aggressively adjusted stereo synthesizer, and then through a surround decoder, you have heard the dialog coming from all the speakers continuously. All localization of the voices to the screen is lost. For this reason, with complete mixes, this is not a desired tool in the transmission path. When used properly, stereo synthesizers can be an advantage for individual mono sources within a Dolby Surround Mix and before encoding. They should not, however, be used excessively when dialog or vocals are part of the mono element.

## 5.12 Dolby Surround Compatible Processors

Several Dolby Surround compatible processors are now available. As with reverb processors, these units offer mixed results. If the unit is substituting for a Dolby Surround encoder, ask yourself “why not use the real thing?” Units that are designed for use with two speakers, 3D audio processors, can produce pleasing results. The secret is to listen to the mix through a Dolby Surround decoder so you know what it will actually sound like to the end user. Encoders that are designed to be used with their own decoders are another issue. The real question is how many consumers are going to be able to hear the mix with the proper decoder. Dolby Surround is the de facto standard matrix surround system worldwide. If a listener has a surround decoder, it most likely is a Dolby Surround decoder.

## 5.13 Mono, Stereo and Dolby Surround Compatibility

Mixing techniques used in Dolby Surround productions are similar to those used in normal stereo productions. Just as you should check mono compatibility of a stereo mix, you should also check mono and stereo compatibility of a Dolby Surround mix.

In most cases, stereo compatibility is not an issue. The mix’s surround element will appear to be outside of the speakers as would an out of phase signal. Notice that the entire mix should not sound out of phase, and there should still be a hard center image.

Mono compatibility is a little trickier. Anything that is in the Surround channel will disappear in mono. This is an asset in some cases and a detriment in others.

In situations like live sporting events, the lack of some crowd information in the mono mix will help those at home listening in mono on a 3-inch television speaker to hear the announcers a little easier. In other applications, the mix may have a critical element in it that is predominately in the Surround channel. For this reason, surround elements should also be present in a front channel, interior panned, so they will be heard in mono, as is commonly practiced in the film industry.

## 5.14 Monitoring

When tracks are being generated for use with other elements like a music mix for film or video, it is often desirable to monitor through a Dolby Surround decoder. These tracks are seldom mixed and encoded before delivery, but are commonly sent as elements on separate tracks (L, C, R, and S). It is important that you know that the elements are acceptable when run through the Dolby Surround encoding system during the final mix or the result can be undesirable. The two most common problems are soundtracks that are very mono in nature or that contain phase information that make them sound surround heavy. The measures required to correct these problems at the final mix session will compromise the mix of the elements, and so these problems should be corrected before the four channel elements are recorded.

## 5.15 Common Pitfalls

Although you may like a surround effect that spins your head, and it may be just what your production needs, repeating the move dozens of times will usually tire the listener. The key to a good surround mix is subtlety. Don’t draw attention to your techniques. The listener should never be distracted from the screen by surround effects. Loud, obnoxious, or out of place effects detract from the production. Keep it fun, but tasteful.

Mixes that are surround heavy will distract the viewer from the on screen action, so do not put too much information in the Surround channel. If you find yourself thinking about what you just heard in the Surround channel instead of paying attention to the action on the screen, you have a problem.. Too much surround information may also make dialog intelligible.

# Chapter 6
# Live Broadcast Applications

## 6.1 Transmission Path Considerations

Because fiber optic and satellite transmission lines are standard services at sports arenas and stadiums, this part of the signal chain is usually repeatable and predictable. Frequency response and headroom are consistent from event to event. Most of the early problems encountered in Dolby Surround broadcasts were related to signal processing added by the station. The two most common causes were *phase chasers* and stereo synthesizers.

## 6.2 Phase Chasers

Dolby Surround uses phase encoding of the Surround channel, so it is entirely possible to have situations where significant out of phase information in the program cannot and should not be corrected, even by well-meaning devices. Phase chasing devices perform two basic functions. One is to correct the small phase errors between two audio channels, such as from azimuth misalignment or drift in analog video tape recorders and cart machines; the other is to correct polarity inversions in one channel relative to the other. When either function occurs in response to Dolby Surround signals, the results can be both unnecessary and unfortunate.

With it becoming the norm to source live feeds or audio from a digital recorder, phase correction is essentially unnecessary, as no azimuth error is introduced. Relative polarity can be easily checked before air time, and this is especially easy when monitoring at the station through an SDU4. If the announcer or dialog comes out the Surround channel, or becomes virtually inaudible in mono, the polarity is inverted! As many people still listen in mono, loss of the announcer either during the game or during a commercial is bound to raise complaints.

## 6.3 Station Limiters

No matter how good your mix is, stations will always try to extract that last quarter dB from the transmitter. This is usually done with a program limiter. In general, avoid using these. But reality dictates that stations need to set these up for minimal effect. Mixers need to mix so that these devices are not required to limit the program material in order to keep the mix in the useful range of the transmitter.

## 6.4 Station Processing

Along with station program limiters, other signal processors may be used, including the Dolby Model 740 Spectral Processor. While there is little that a mixer can do while on location to defeat the use of these units, if they have been properly set up, without too much radical signal processing, they should not pose a problem.

## 6.5 Headroom

The amount of available headroom depends on the final distribution medium . In broadcast transmission applications 6 dB of headroom above reference level is the usual limit. For transmission paths between venues and satellite uplinks, 10 dB may be available. VHS tape hi-fi tracks have about 12 dB of headroom. CDs, laser discs, and digital video recorders have 20 dB of headroom. Know what headroom is available and stay within the boundaries. Failure to do so will create unwanted side effects. The headroom of the encoder and decoder are sufficient for any of the requirements above.

## 6.6 Stereo Synthesizers in Transmission Paths

Although stereo synthesizers, used to make mono programs *stereo*, are less common now than they were when stereo TV first became a reality, they are still in use. They must not be used in the transmission path of a Dolby Surround encoded soundtrack after the encoding process. Use of these devices will destroy all surround encoding. Also, although the effect may sound all right to some people in stereo for a mono program, the result of any stereo synthesized program when decoded through a Dolby Surround decoder is nothing short of obnoxious.

# Chapter 7
# Video Games and Multimedia

## 7.1 Introduction

Dolby Surround has become a popular tool for video game and multimedia developers. There are several ways to include Dolby Surround in a video game, and each depends on the game's platform.

Control of the sound's spatial position in response to the game player's input is often desired. There are several techniques that can be used to achieve this in the game itself. The degree of complexity and resulting quality varies, and so the technique should be chosen based on the intended effect.

## 7.2 Normal Dolby Surround Encoding

The easiest method is to simply encode all of the audio files for the game in Dolby Surround. This will give you a pleasing, but non-interactive soundtrack. To accomplish this, mix the project as you would any other type of project and save the sounds as stereo files with the game. Dolby Surround encoded stereo files can be handled like any other stereo file in the game. The files can be downsampled to 16 bit, 22 kHz stereo without destruction of the encoding within the file.

## 7.3 Polarity Inversion

Another method is to emulate Dolby Surround. This is common in console games. This method lets you place sounds at each of the four cardinal points: Left, Center, Right and Surround. This is an approximation of Dolby Surround and is limited in its capability. For simple console games where sound file space is at a premium, this technique produces an adequate Dolby Surround effect.

In-game sound effects need to follow the action on the screen. Sound placement can be handled within the game itself, using only basic controls. The front cardinal points, Left, Center, and Right, can be encoded by panning like any conventional stereo signal. Simple channel switching during game play allows the sound to be output from the Left, Right or Center (both channels). If the game provides a variable balance control function, the sound may be smoothly panned across the front or positioned between channels.

The simple frontal positioning may be adequate for many sounds that need run-time control. For more realism, placing sounds in the Surround channel may be desired as well.

Sounds appear in the Surround channel when the Lt/Rt signals into the decoder are equal in level (much like a center signal), except one of the signals is inverted, commonly called 180° out-of-phase, relative to the other.

By adding an inverter, multiplication by -1, to the game audio tool kit, it is possible for a sound to be placed at any of the four cardinal points. If the level of one of the two output signals can be attenuated, it is possible to move the sound across the front or down either side toward the Surround channel. This degree of sound positioning is often adequate for many situations in game play. However, it does not allow for either interior sounds or center-to-surround pans. To achieve these effects, phase encoding is needed.

If the game audio playback software allows for the creation of small time delays or certain filter or pitch-shift functions, these may be suitable for creating signals that appear to spread wider or otherwise occupy the interior space.

## 7.4 Phase Encoding

To increase sound positioning capability in more advanced games, Dolby has prepared a simple C-code program for surround encoding during game play called the Dolby Phase Positioner. This simplified encoding utility is comprised of two sections: a phase shifter and a surround positioner. The shifter takes any mono signal and creates two new audio signal components, F (Front) and S (Surround), which are phase shifted relative to each other. If the game's audio engine can output multiple streams of audio that are pre-panned to L, R, C, and S, the Dolby Phase Positioner can also encode these and mix them to a Dolby Surround compatible signal.

The positioner moves the F input signal along the front axis in response to the L/R input control. It moves the S input signal along the surround axis in response to the F/S input control. This x/y coordinate system allows the game to position the sound anywhere in the Dolby Surround sound field.

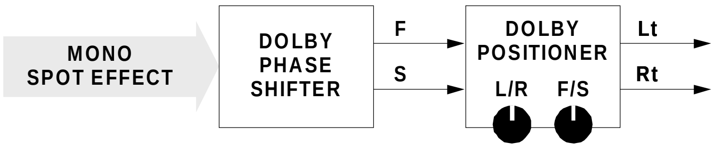


**Figure 7-1 Phase Shifter and Positioner**

The phase shifter is deliberately simple to minimize the impact on game speed. Since the shifter is needed only when a sound is placed or panned through the interior space, it need not run at other times. If the phase shifter is too much of a processing burden to run during game play, or if the sound quality is not deemed good enough for some reason, the positioner can also work with signals preprocessed in what is called game mode.

## 7.5 Dolby Surround Game Mode Encoding

This method involves encoding the main tracks with Dolby Surround and then encoding certain effects tracks with game mode encoding. This option gives you basic audio with Dolby Surround encoding as well as the ability to position additional sounds as the game is played. In this mode, the final encoding of these sound effects is done by the game itself.

Dolby Surround Game Mode is designed to assist computer game developers with creating sound effects and elements that can be used within the game and spatially placed within the sound field by the game while being played.

This is a mode of the Dolby Surround encoder itself: A mono input signal, the audio that will be positioned by the game, is processed by the game mode encoder to produce a two-channel output signal having the same F and S components as with the phase shifter just described. During game play, this two-channel signal may be moved using the positioner as described above.

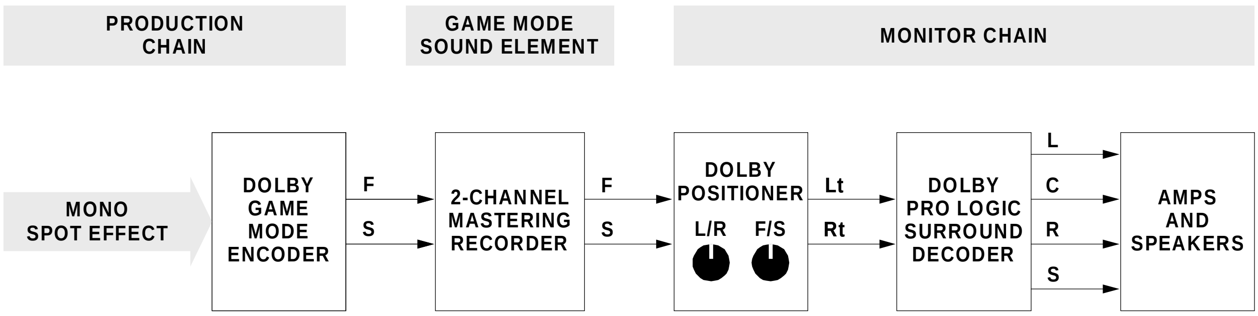


**Figure 7-2  Signal Flow for Game Encoding**

The resulting signal has the same audio processing as other true Dolby Surround encoded signals, and needs no extra phase shift processing during game play. The main disadvantage is that this mono element now occupies twice the audio storage space a mono element would in the delivered program.

## 7.6 Modification Principle

Game mode encoding is accomplished by modifying a Dolby Surround encoder to remove the surround signal from the phase shift combine for the Left channel output of the encoder. The input signal to be encoded is delivered to both the Left and Surround channel inputs of the encoder.

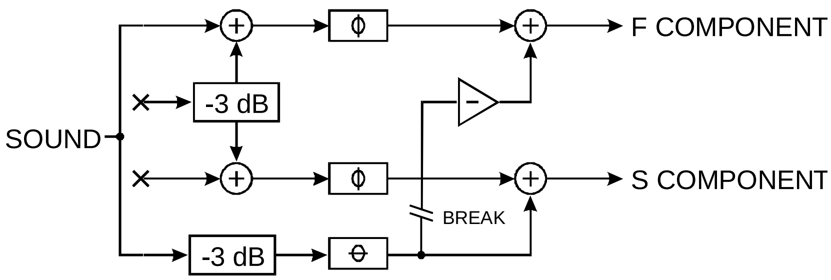


**Figure 7-3  Dolby Surround Game Encoding**

The left output (labeled Lt) of the encoder becomes the front information (F) and the right output (labeled Rt) of the encoder becomes the rear (S) information. If you are using the Game Mode encoder in Dolby Surround Tools, these outputs are Left and Right, respectively.

Once the F and S signals are prepared, they are recorded in two channels of the game soundtrack. To create the final surround encoded outputs, the F and S components are blended together during game play using the simple mixer shown in *Figure 7-4*.

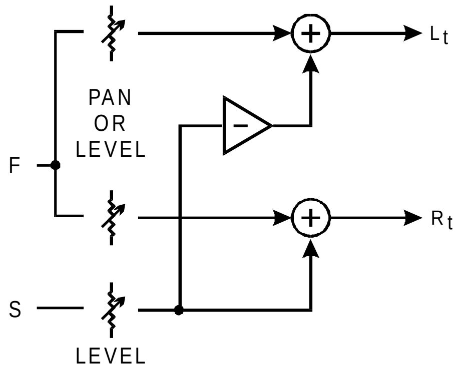


**Figure 7-4 Game Mixer**

## 7.7 Application Information

Game mode encoding is only useful for the generation of sound files for use in video games and the like. The output of the encoder can not be monitored without additional circuitry or signal processing which is normally part of the video game. The output of the encoder should look like a near circle, an ellipse, on a phase scope as in *Figure 7-5*.

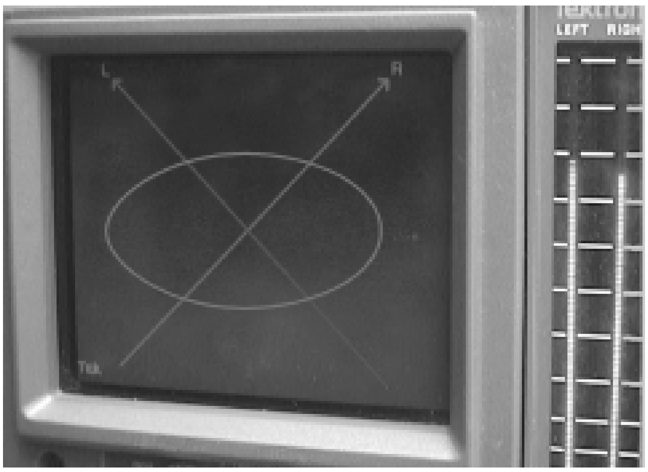


**Figure 7-5 Waveform of Game Encoded Signal**

When measured for level, the Left channel output (front information) should be the same level as the input to the Left channel. The Right channel output (surround information) should be 3 dB lower in level than the Surround channel input.

## 7.8 SEU4 Game Mode Alignment

If the encoder has been aligned for normal operation, no changes to alignment will be necessary for game mode. If the encoder has not been aligned, you can either set it up following the normal operation, with the unit in the normal mode, or follow the procedure below which is valid only for game mode.

1. Make sure the unit is in the game mode. To determine this, look at the Surround Active LED on the front of the encoder, see *Figure 3-10*. If the unit is modified for game mode and the LED is on, the unit is in the normal mode. If the LED is off, the unit is in the game mode.
2. Apply 1 kHz at 0 dB reference level to the left input.
3. Adjust the left input level to light both green LED’s.
4. Adjust the Lt output for unity gain or as appropriate for your situation.
5. Remove the 1 kHz tone from the Left channel and apply it to the surround input.
6. Adjust the surround input to light both green LED’s.
7. Adjust the Rt output for -3 dB relative to unity gain or as appropriate for your situation.

**Alternate Method:**

Put the 1 kHz tone into the right input, adjust the input to light both LED’s and adjust the Rt output for unity gain or as appropriate for your situation.

## 7.9 Testing Game Mode Encoding with an Audio Console

Because the output of the encoder in game mode cannot be directly monitored with the decoder, additional circuitry or processing is required to properly assemble the signals. In most circumstances, this is accomplished within the game. However, it is possible to take the two encoded signal outputs from the encoder, run them through the console and mix them together to check for compatibility through the decoder.

To do so, connect the left (F information) output to a console fader, the right (S information) output to a second fader, and a polarity inverted version of the Rt output to a third fader. The polarity inversion may be accomplished by using the phase invert switch for the applicable input module on the console, or by wiring the input connection with a reverse wired connector or patch cable.

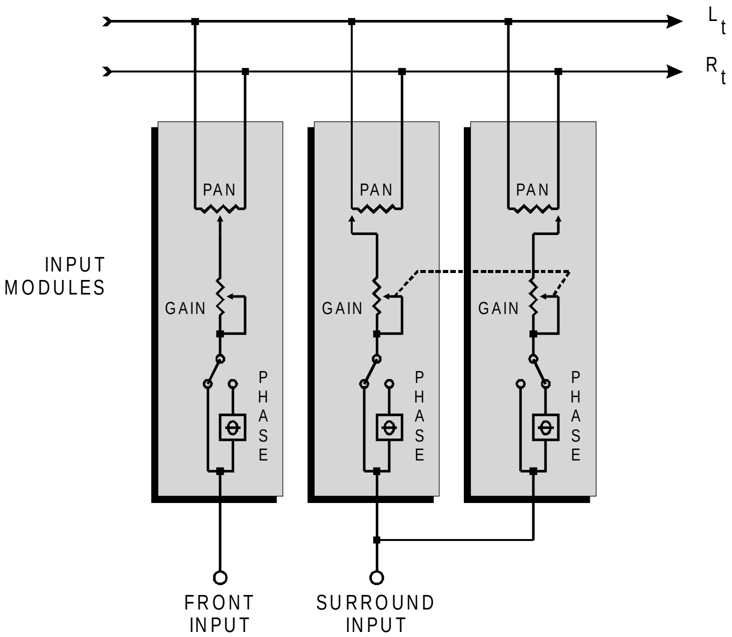


**Figure 7-6 Positioner Function Via Audio Console**

The left (F) signal is panned from left to right for the front channel information. The right (S) information that is in phase should be assigned to the Left channel, the Rt information that is inverted polarity should be assigned to the Right channel. The Left and Right channels from the console then feed the SDU4 decoder. By bringing up both the inverted and non-inverted polarity S signals together, surround information will appear. Bringing up the front fader will affect the front level. By bringing up the front fader while bringing down the 2 back faders, pans from back to front will occur. Reversing this action will pan from front to back.

The following table summarizes the basic encoding methods discussed above. No one method is necessarily perfect for all cases. The best judge is the final result. Please contact Dolby Laboratories in case questions arise.

**Table 7-1 Dolby Surround Game Encoding Options**

| Game Characteristic | Studio Encoder | Game Mode | Phase Shifter/ Positioner | Polarity Inversion |
|---|---|---|---|---|
| **MIPs Impact** | None | medium | High | low |
| **Positioning Range** | n/a | very good | very good | mainly surround |
| **Sound Quality** | very good | very good | Good | very good |
| **Delivery Impact** | None | 2x sound file | None | none |

## 7.10 Game Playback

A game may use any one or all of the surround encoding options described and still be a valid Dolby Surround game as long as the end result is consistent with the general quality found in other such games.

*Figure 7-7* shows how the various sound elements may join together in the final game. The final audio output is a complete surround mix in the Lt/Rt encoded form, which is output from the sound card. It is then the task of the Dolby Pro Logic decoder to extract the multichannel sound field for reproduction over several speakers, or to be further virtualized for playback over a conventional pair of speakers.

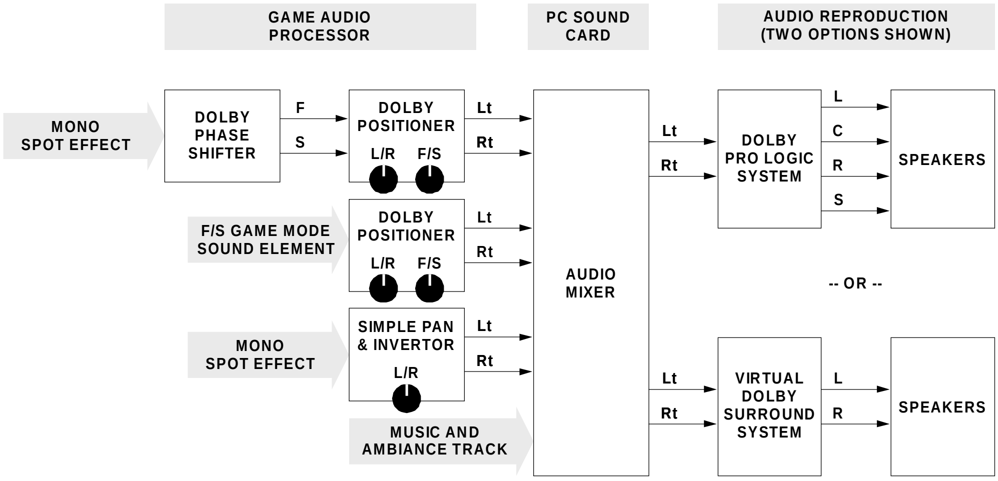


**Figure 7-7 Game Audio Creation and Reproduction**

# Chapter 8
# Theory of Operation

## 8.1 Encoder

*Figure 8-1* is the block diagram of a Dolby Surround encoder.

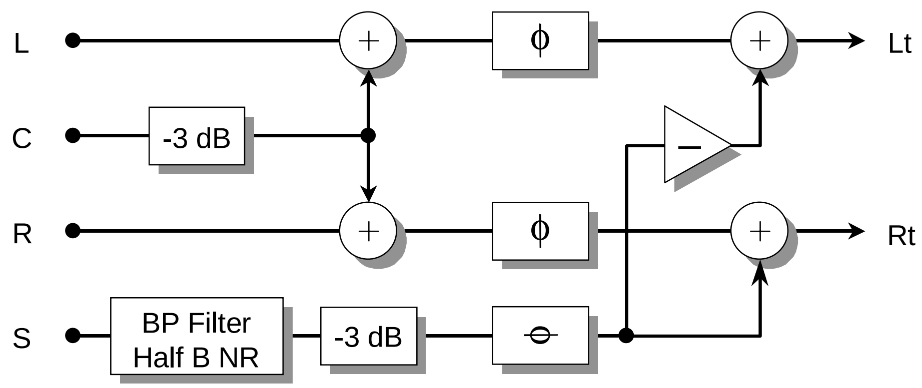


**Figure 8-1 Dolby Surround Encoder**

The encoder adds the center channel input C, attenuated by 3 dB, to the left and right input signals, L and R. The results are sent through all-pass networks providing frequency-dependent phase-shift and fed to the left and right total outputs, Lt and Rt.

The surround input S is attenuated by 3 dB, bandpass filtered from 100 Hz to 7 kHz, and passed through a Dolby B-type encoder modified to produce 5 dB of noise reduction rather than the normal 10 dB. The result is sent through a separate all-pass network and added to the Rt output and subtracted from the Lt output. The S input therefore, yields two surround signals of opposite polarity from the Lt/Rt encoder outputs.

All processing in the surround path contributes to the total degree of phase shift for that channel. The all-pass networks are designed so that over the range of the surround bandpass filter, the phase shift of the surround path output always lags that of the left and right by as close to 90 degrees as pratically possible. All-pass networks with this property have large frequency-dependent phase lag. Thus for instance at 1 kHz, the left and right paths through the Dolby Model SEU4 give phase shifts of roughly –550 degrees, while the surround path, measured at the right total output, has about 90 degrees more lag (approximately –640 degrees total).

## 8.2 Decoder

The Dolby Surround decoder detects the equal amplitude in phase (center) and equal amplitude inverted (surround) signals. These signals, combined with the Left and Right signals during encoding, are decoded by the steering matrix based on which signal is dominant at any time. The Center channel information can either be fed from the Center channel input of the encoder or sent to both the Left and Right inputs equally. The decoder does not differentiate between the two methods. For this reason, signals panned to the center of the left/right bus will appear in the Center channel output after decoding, not the Left and Right channels.

*Figure 8-2* is a block diagram of the Dolby Surround Pro Logic decoder.

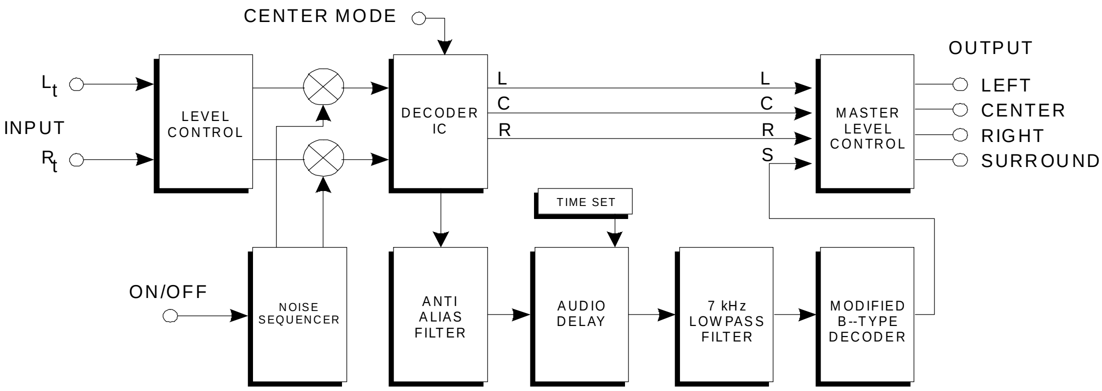


**Figure 8-2 Dolby Surround Pro Logic Decoder**

In the SDU4 decoder the input signals are first fed to the input level control section to match the decoder reference level to the encoder reference level. The noise sequencer, which generates the calibration noise used during the speaker system setup, is added to the output of the input level controls and is fed to the input of the matrix steering decoder. The matrix steering decoder routes the Center channel signal and controls the level of each of the four matrix outputs to increase the separation between channels whenever possible. The Left, Center, and Right signals are then sent to the master volume control. The Surround channel is first sent to an anti-aliasing filter, an adjustable length digital delay, a 7 kHz low pass filter, and a Dolby B-type noise reduction decoder, which has complimentary modifications to match the B-type noise reduction encoder used during the surround encoding process. The output of this signal chain is then sent to the master volume control. The output of the master volume control is the output of the decoder.

# Chapter 9
# Miscellaneous Information

## 9.1 Contacting Dolby Laboratories

In addition to its headquarters in San Francisco, Dolby has several other offices around the world. All offices are equipped to provide information on soundtrack production and encoding.

You may contact Dolby from anywhere in the world via the following e-mail addresses:

| Address | Use |
|---|---|
| info@dolby.com | General information and inquiries |
| tsa@dolby.com | To apply for a Dolby trademark agreement (TSA) |
| info@dolby.com | Questions on audio encoding for DVD |
| info@dolby.com | Questions on multimedia applications |

A wide variety of technical and trademark information can be found on Dolby's web site at www.dolby.com.

Below is information on local Dolby offices. Please contact the nearest office for assistance.


### San Francisco Headquarters

Dolby Laboratories
100 Potrero Avenue
San Francisco, CA 94103-4813
**Phone** 415-558-0200
**Facsimile** 415-863-1373

### Los Angeles

Dolby Laboratories
3375 Barham Boulevard
Los Angeles, CA 90068-1446
**Phone** 323-845-1880
**Facsimile** 323-845-1890

### New York

Dolby Laboratories
1350 Avenue of the Americas
New York, NY 10019-4703
**Phone** 212-767-1700
**Facsimile** 212-767-1705

### England

Dolby Laboratories
Wootton Bassett
Wiltshire SN4 8QJ England
**Phone** (44) 1793-842100
**Facsimile** (44) 1793-842101

### Shanghai Liaison Office

Dolby Laboratories Representative Office
7/Fl. Hai Xing Plaza, Unit H
Rui Jin Road (S)
Shanghai 2000023 China
**Phone** (86) 21-6418-1015
**Facsimile** (86) 21-6418-1013

**Japanese Liaison Office**

Dolby Laboratories International
Services, Inc.
Japan Branch
Fuji Chuo Building 6F
2-1-7, Shintomi, Chuo-ku
Tokyo 104-0041 Japan
**Phone** (81) 3-5542-6160
**Facsimile** (81) 3-5542-6158

**CFE PRO**

Roppongi Office
Saski Building 18-9
Roppongi 3 Chome
Minato-ku, Tokyo 106-0032 Japan
**Phone** (81) 3-583-84515
**Facsimile** (81) 3-589-0272

## 9.2 Software Identification and Trademark Usage

Dolby Laboratories encourages use of the Dolby Surround trademark to identify soundtracks that are encoded in Dolby Surround. This is an effective way to inform listeners of the soundtrack format, and the use of a standard logo promotes easy recognition in the marketplace. However, like any trademark, the Dolby Surround logo may not be used without permission. Dolby Laboratories therefore provides a royalty-free Trademark and Standardization Agreement (TSA) for companies that wish to use Dolby trademarks. This agreement must be signed by the company that owns the program material being produced. Recording studios or production facilities that provide audio production, encoding, or manufacturing services for outside clients generally do not require a trademark license. However, we do ask that these facilities refer their clients to us for trademark licensing information.

If you would like to use the Dolby Surround logo, you can apply for a Dolby Trademark and Standardization Agreement (TSA) by sending e-mail to tsa@dolby.com or by contacting Dolby Laboratories at any of the locations given in *Section 9.1 Contacting Dolby Laboratories*. When sending written requests please indicate that you would like a Dolby Surround trademark license and include your name, your company name, mailing address, and the type of media on which your soundtracks will be distributed, such as CD, laser disc or VHS.

For detailed information on Dolby trademark licensing, please refer to the document Use of Dolby Trademarks on Audio and Video Media, available on the Dolby web site at www.dolby.com. We are also planning to make our license application form available on-line, so check the Dolby web site in the coming months for the on-line version of the Media Licensing Questionnaire.

If you are already a Dolby licensee and would like more information on trademark use, please contact Dolby Laboratories. We are always happy to review artwork and assist with the proper use of our trademarks. Information on trademark licensing plus instructions for using the Dolby Surround trademark and marking audio features on DVD can also be found on the Dolby web site.

## 9.3 Dolby Surround Consultants

Dolby Laboratories has consultants available to assist with setting up rooms, checking calibrations, and mixing. Consultants are available for a fee for either half or full days. Charges for engineering services do not include travel to and from the studio. If the facility is outside of the local area of one of our offices, travel, hotel and meals will be billed at actual cost.

## 9.4 Dolby Surround Software Lists

To inform consumers of programs, CDs, games and videos available with Dolby Surround encoding, Dolby Laboratories maintains listings on the Dolby Web Page (www.dolby.com) and in print. To keep these lists current, studios and engineers producing Dolby Surround encoded programs are encouraged to inform us of their titles for inclusion on these lists. Send program information to info@dolby.com.