

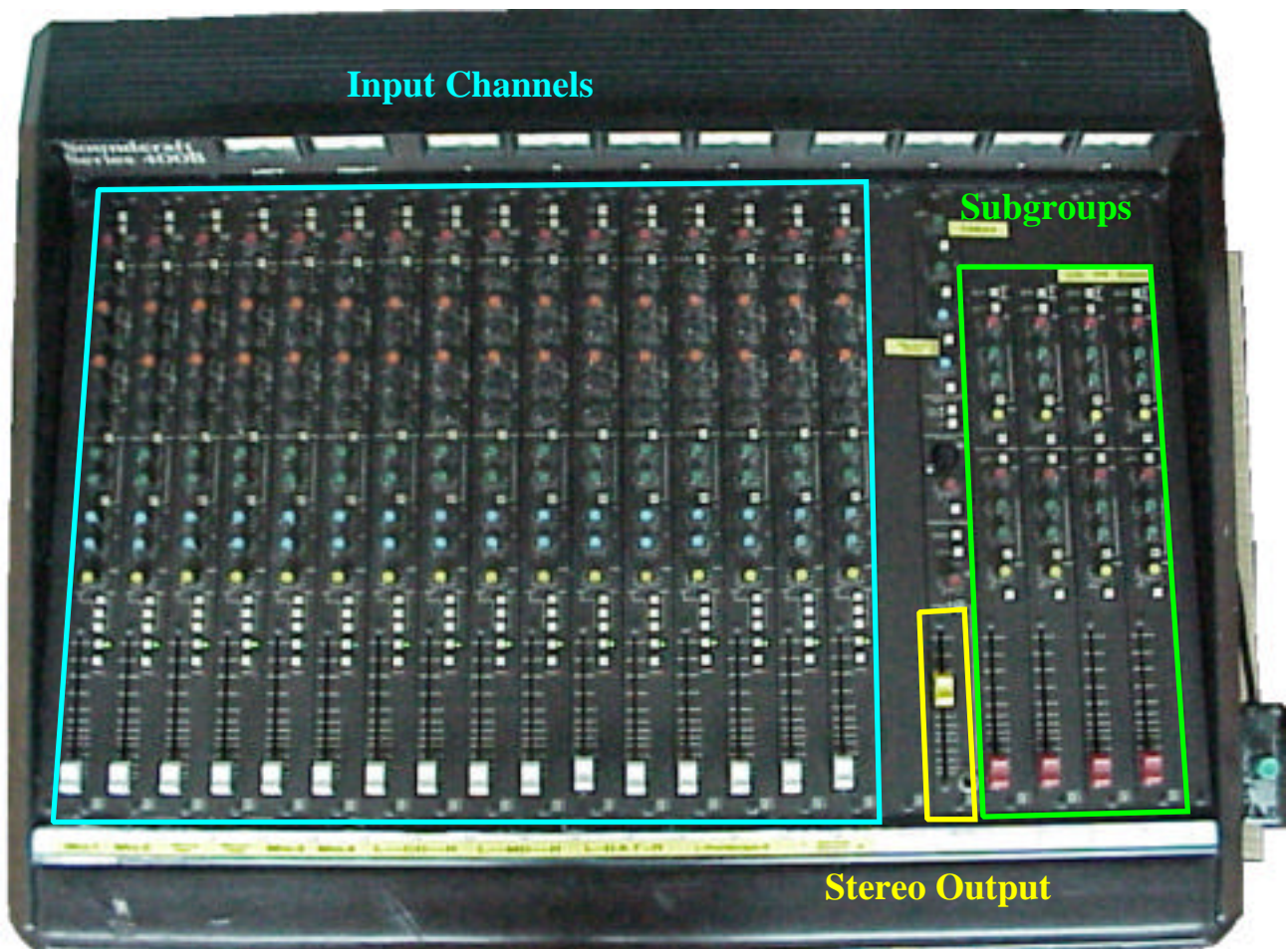
# Studio Notes

## Studio Three – The Control Room

The control room of studio three is where the performance in the sound isolation booth is recorded. It essentially consists of four areas, the mixer, recording devices, effects processors and video playback.

It is the job of the recording engineer to ensure that a good recording is made of the performance in the sound isolation booth.

### The Mixer



The mixer in the control room is a Soundcraft Series 400B mixer. The mixer is where the outputs from the microphones in the recording room are routed to. From the mixer, the signal can be recorded to any of the devices in the rack, with the exception of the video recorder. Any of the recording devices can be played back via the mixer through the JBL Control 5 monitors.

The mixer consists of 16 input channels. They have the following plugged into them:

Channel Number	Input from:
1	Mic 1 (via the VC7)
2	Mic 2 (via the VC7)
3	Mic 1+2 Mid (from the VC7)
4	Mic 1+2 Side (from the VC7)
5	Mic 3
6	Mic 4
7	CD left
8	CD right
9	MiniDisc left
10	MiniDisc right
11	DAT left
12	DAT right
13	Soundscape left
14	Soundscape right
15	Reverb Return left
16	Reverb Return right

## The Mixer Input Channels

The input channels form the largest area of the mixer. They may look daunting at first, but closer inspection shows that there are 16 repeats of the same set of controls. Figure 1 shows the layout and function of one of these channels.

### Input Section

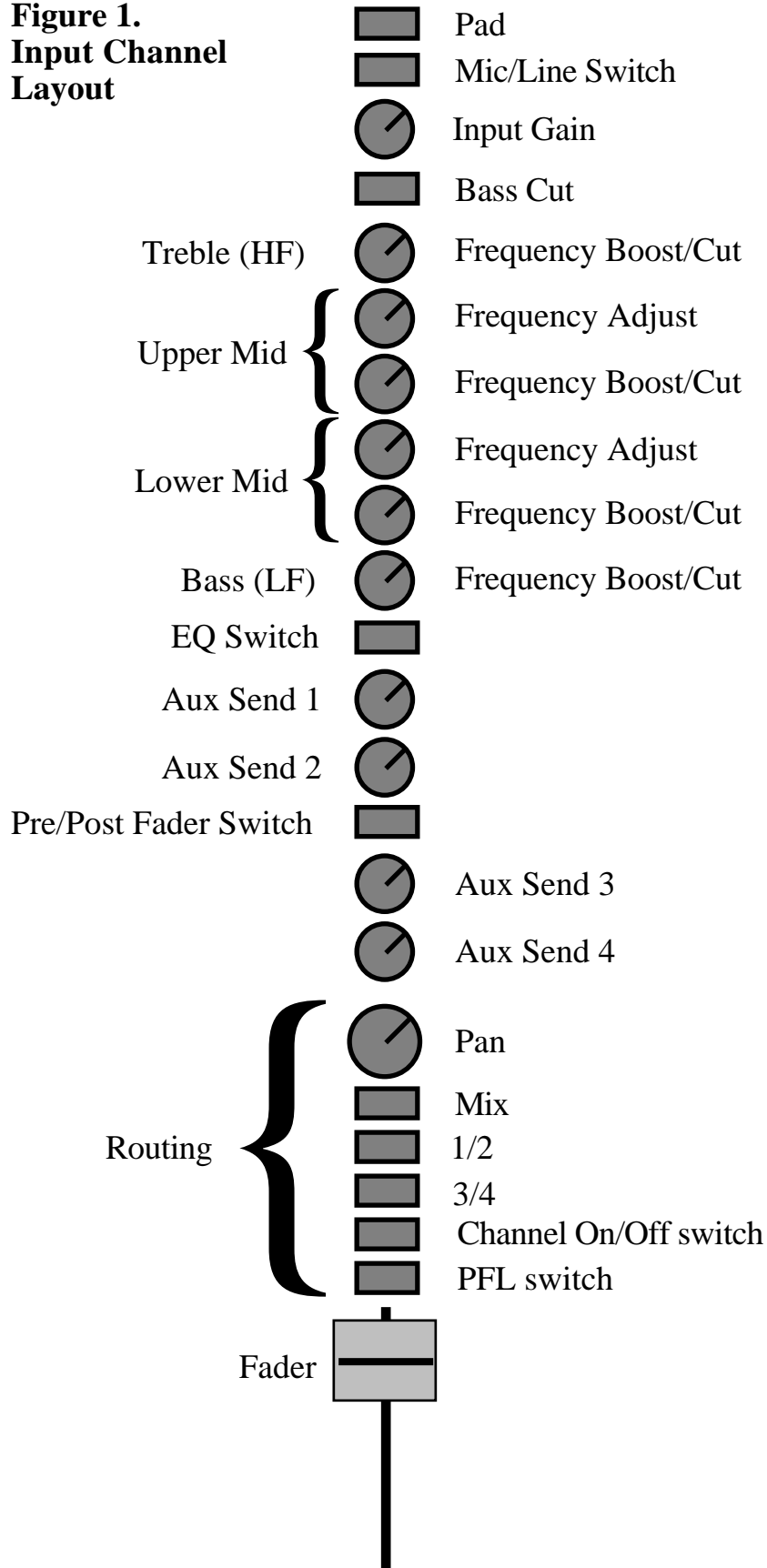
**Pad:** the pad button is for changing the input signal level. (It reduces the gain by -20dB and is used when the input level is so loud that turning the input gain dial to its lowest setting fails to lower the signal enough to prevent distortion.)

**Mic/Line Switch:** the mic/line switch is used to select between the microphone and line level inputs as the channel input source. The switch should be in the upper position for microphone inputs and in the lower position for line level inputs.

**Input Gain:** the input gain is for adjusting the level of the input into the channel to give the loudest distortion free signal. The fader should be set to 0dB when making adjustments to this control.

**Bass Cut:** the bass cut switch is used to cut very low frequencies, e.g. rumble and microphone noise. When the switch is in the lower position the bass cut is active, it is disabled when the switch is in the upper position.

**Figure 1.  
Input Channel  
Layout**



## Equalisation Section

The equalisation section of the mixer is used to change the tonal quality of the incoming signal by boosting or cutting different sections of its frequency range. The mixer has semi-parametric equalisation which means that there is a certain amount of choice over which frequencies you can adjust.

**Treble (HF) Boost/Cut:** the treble control is the part of the equalisation section that controls the high frequencies of the incoming signal. It allows you to boost or cut the high frequency range by up to 15dB.

**Upper-Mid Frequency Adjust:** this control is used to select the centre frequency that is boosted/cut in the range of 600Hz to 10kHz.

**Upper-Mid Frequency Boost/Cut:** this control is used to boost or cut the level of the signal in the frequency range set by the upper-mid frequency adjust control by up to 15dB.

**Lower-Mid Frequency Adjust:** this control is used to select the centre frequency that is boosted/cut in the range of 150Hz to 2.4kHz.

**Lower-Mid Frequency Boost/Cut:** this control is used to boost or cut the level of the signal in the frequency range set by the lower-mid frequency adjust control by up to 15dB.

**Bass (LF) Boost/Cut:** the low frequency control is the part of the equalisation section that controls the low frequencies of the incoming signal. It allows you to boost or cut the low frequency range by up to 15dB.

**EQ Switch:** the EQ switch is used to enable or disable the equalisation section of the channel. It is enabled when the switch is in the lower position and disabled when in the upper position. Normally you would not want to EQ signal when recording.

## Auxiliary Bus Section

**Aux Send 1 and 2:** in this set-up, these two aux sends are used for fold back to the studio speakers.

**Pre/Post Fader Switch:** this switch is used to hear the signal on the auxiliary busses 1 and 2 when the fader is down. The signal is heard pre-fader in the down position and post fader in the up position.

**Aux Send 3 and 4:** in this set-up, these auxiliary busses are used to send part of the signal to a reverb unit. Aux 3 sends to the left channel and aux 4 to the right channel of the effects processor. The level set on these controls determines the amount of signal that is sent to the processor. The effects processor is returned to the mixing desk on channels 15 and 16.

## Routing Section

**Pan:** the pan control is used to direct the channel signal to the left or right of the main out or subgroup channels. In a stereo pair of channels, e.g. the input from the CD player, you would have these panned so that the odd numbered channel was hard left and the even channel hard right.

**Mix Switch:** the mix switch is for routing the channel signal through to the main output channels. This is the routing to use in this studio.

**1/2 Switch:** the 1/2 switch is used for routing the channel signal to sub-groups 1 and 2.

**3/4 Switch:** the 3/4 switch is used to route the channel signal to sub-groups 3 and 4.

**ON Switch:** this switches the channel on and off. It is turned on in the down position, (a green light lights up), and off in the upper position.

**PFL Switch:** The prefade switch acts as a solo switch, it lets you monitor a channel whilst switching off those whose PFL switches are not depressed. This switch should not be used instead of the On switch, it prevents signals appearing at the main output so recordings would not be able to be made onto the DAT.

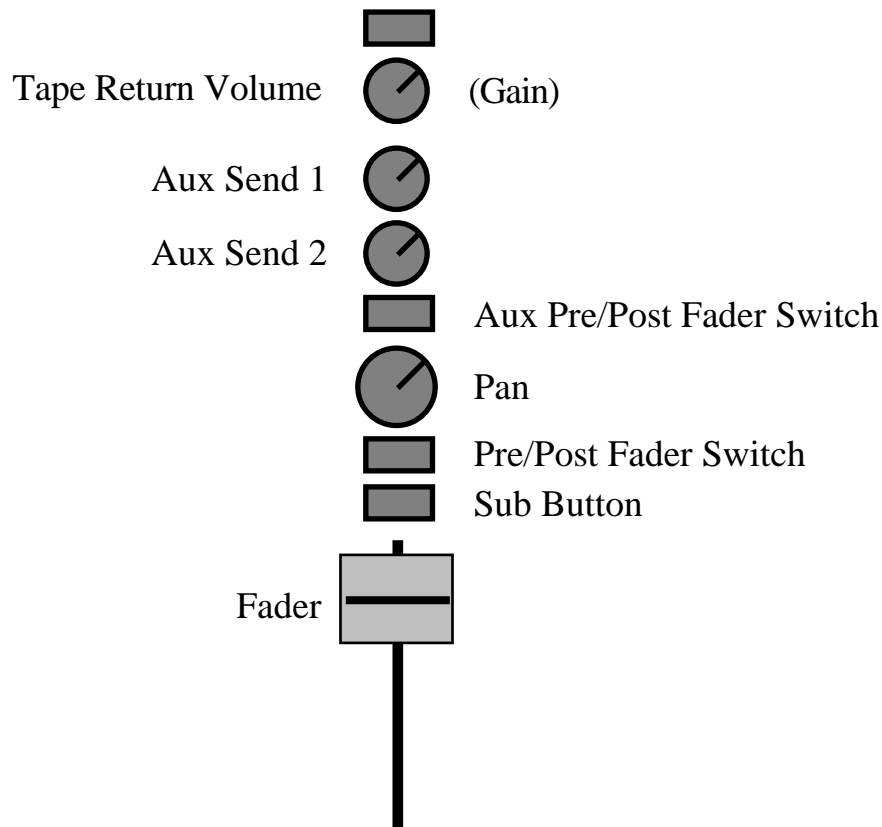
**Fader:** the main fader is used to control the level of the channel signal in the mix. The fader should be at 0dB when setting the input gain to get the optimum signal level.

## Sub-Groups

Input audio channels can to be routed through to sub-group channels, but in the setup for this studio, the subgroups are not used except to hear the tone. Subgroup channels usually form a kind of "mini-mixer" that lets you mix your input channels down into four subgroups, useful if you are mixing down onto a four track recorder. Each subgroup channel has the same controls.

**Tape Return Volume (Gain):** this control is used to adjust the gain of the signal coming into the subgroup

**Aux Send 1 and 2:** these controls are used to set the amount of signal to be sent to the appropriate auxilliary busses.



**Figure 2.**  
**Sub-Group Channel Layout**

**Aux Pre/Post Fader Switch:** this is used to determine whether you hear the signal being sent to auxiliary busses 1 or 2 before or after the fader. In the upper position it is post fader and in the lower position it is pre fader.

**Pan:** this dial is used for routing the signal from the subgroup to left or right.

**Pre/post fader Switch:** is used to monitor the subgroup channel signal before or after the fader. In the lower position it is before the fader and in the upper position it is after the fader.

**Sub Button:** leave this button in the upper position, or risk the chance (certainty) of feedback.

**Fader:** this is used to balance the output levels between the four subgroups.

### **Other Controls on the Mixing Desk**

**Tone:** this button is used to produce a 700Hz or 10 kHz tone. This is used for tape head aligning and also for recording a reference tone for master recordings. Normally the reference tone is recorded at  $-4.0\text{dB}$  on an analogue recorder or  $-18.0\text{dB}$  on a digital recorder. In this setup, the tone is routed through the subgroups, so to hear it

you will need to set a subgroup fader to the 0dB position and turn up its input volume control. Check the VU meters to see that the levels are at the appropriate level. If you are recording the material you record in your studio session to an analogue source, e.g. a beta online session, at a later date, then it is wise to record about 30 - 60 seconds of test tone at the start of your DAT tape at -18dB so you can use this as a reference tone to set up record levels to -4dB on the analogue recorder.

**2 Track Replay:** this button is not connected and therefore not used in this setup.

**Mono Check:** this button is used to check for phase errors between the left and right channels of a stereo mix. If, with the button depressed, the signal weakens or disappears, there is a phase problem. If pressing the button does not reduce the level of the signal, there is no phase problem. This is worth checking if you are using spaced microphone techniques to record stereo sound.

**Auxilliary Master Send:** each of the auxilliary busses has a master send level control to set the level of the signal that is sent to it.

**Talk Back:** this button is used for talking to the performers in the audio room, either via headphones or the monitor. The red dial above the microphone is used to set the level of signal sent to the studio. Check to make sure there is a microphone in the socket! There is a slight danger of audio feedback via microphones and monitor speaker in the studio and the speakers and talkback microphone in the control room. To help minimise this risk, have the microphones pointed away from the monitor speaker and a reasonable distance from it. In the control room, don't leave the talk back button depressed whilst the performer is performing and keep the return level relatively low.

**Monitor Volume:** this is used to set the level of the playback speakers in the control room.

**Master Fader:** this fader, the yellow one, is used to set the level of the main output of the mixer, hence the level of the signal going to the DAT, Soundscape and the MiniDisc recorder.

**Cue Light:** at the side of the mixer is a big green button, this activates a light in the studio and is used as a silent indicator that recording has started.  
The mixing desk has a phantom power supply for condensor microphones.

**The rack in the control room contains recording, playback and processor units**



- Soundscape Unit
- PPS100 Sync Device
- MidiVerb II Reverb Unit
- Dynamics Processor
- Joe Meek VC7
- VHS Player
- CD Player
- MiniDisc Recorder
- DAT Recorder

## **The Recording Devices**

The control room has several recording units.

Digital Audio Tape (DAT)

MiniDisc recorder

Soundscape

The output of the mixing desk is routed to all of these, so to record to a device simply set it up to record, set the levels and record.

You cannot record to the CD Player (!) or the VHS machine.

## **The Playback Devices**

In addition to the recording devices mentioned above, all of which play back, there is a CD player and VHS player in the studio rack.

To playback any of the devices, (except for the VHS player), through the studio monitors, turn up the aux send for the channels on which the device appears. Use aux



send 3 for odd numbered (left) channels and aux send 4 for even (right) channels. Adjust the Mon Vol pot to adjust the playback level of the speakers.

To play back the VHS player, use the subgroups 3 and 4 at the top right of the mixer. Use the red volume knobs to set the playback volume and the yellow pan knobs to set the panning. If you are using a timecode striped tape, then reduce the level of the left hand channel to zero, which will carry the SMPTE time code, and pan the right hand channel, which carries any sync sound, to the centre and set it's volume to a reasonable level. If your tape simply has recorded material on it, for example if you want to record an instrument part to a pre-recorded track, then pan the two tracks hard left and hard right to preserve the stereo recording.

## **How To Record**

### **Mono Recording**

First you need to set up your microphone in the sound booth and plug it in to one of the microphone sockets in the unit by the window. Use sockets 1 or 2 if you want to use the Joe Meek VC7 preamplifier, and this is recommended, or sockets 3 or 4 if you want to send the signal straight to the mixing desk.

In the control room, turn the input gain down on the channel that corresponds to the socket you selected and put the output fader up to 0. Set the pan dial to centre. Now get the performer to perform for a little while so that their level of performance settles to a level and turn up the input gain until the VU meter reads about 0. It's all right for the levels to go above this, but they should not be constantly above it. Since the sound booth is acoustically dead, there shouldn't be any noise problem with this level. Now you have set the best signal to noise ratio, you should only use the output fader to adjust level from now on.

To record, set the record level on the target device and record...

### **Stereophonic Recording**

First, set up the mixer channels for level as for monophonic recording, except this time you need to make sure that the levels for each channel is the same and the pan pots are set hard left for the odd numbered channel and hard right for the even numbered channel.

If you are using the mid-side recording technique, then you need to use the Joe Meek VC7 unit. You will have plugged the microphones into sockets 1 and 2 in the sound isolation room and these are routed through to the Joe Meek unit in the rack. This unit converts the two microphone signals into a stereo signal by passing them through an A+B and A-B matrix which uses a process of phase cancellation between the outputs of each microphone to produce a stereo image.

The output of the matrix from the VC7 is routed to channels 3 and 4 on the mixing desk. So if you are using the mid-side technique for recording, you should be recording the outputs from channels 3 and 4 on the mixing desk, since these are the proper left and right signals as interpreted via the matrix in the VC7.

If you want to, you can record the mid and side microphone signals direct to the recording device and process them in soundscape at a later time.

If you are using spaced microphone or coincident microphone stereo recording techniques, then the output from the matrix on the VC7 is actually a mid-side signal, the matrix converts the stereo signal into what the mid and side microphones would have picked up! In this case, to record a stereo signal, use the outputs from channels 1 and 2 on the mixing desk, these are the signals coming direct from the microphone.

## **Synchronised Recording On Soundscape**

You can record to Soundscape when it is synchronised to a striped VHS tape in the following way:

Cue up the VHS tape, allowing at least 20 seconds of pre-roll time before the action you want to synchronise to.

Check the time code on the VHS tape and make sure that the SMPTE Offset in Soundscape is set accordingly. For a program that starts at 10.00.00.00, the SMPTE offset would typically be set at 09.58.00.00. (**settings menu > SMPTE Offset**)

Set the input source in Soundscape to analogue (**settings menu > input > analogue**)

Set up two (or one) record tracks in Soundscape, using the left and right locators to set the record time and the microphone icons to set the track between the locators.

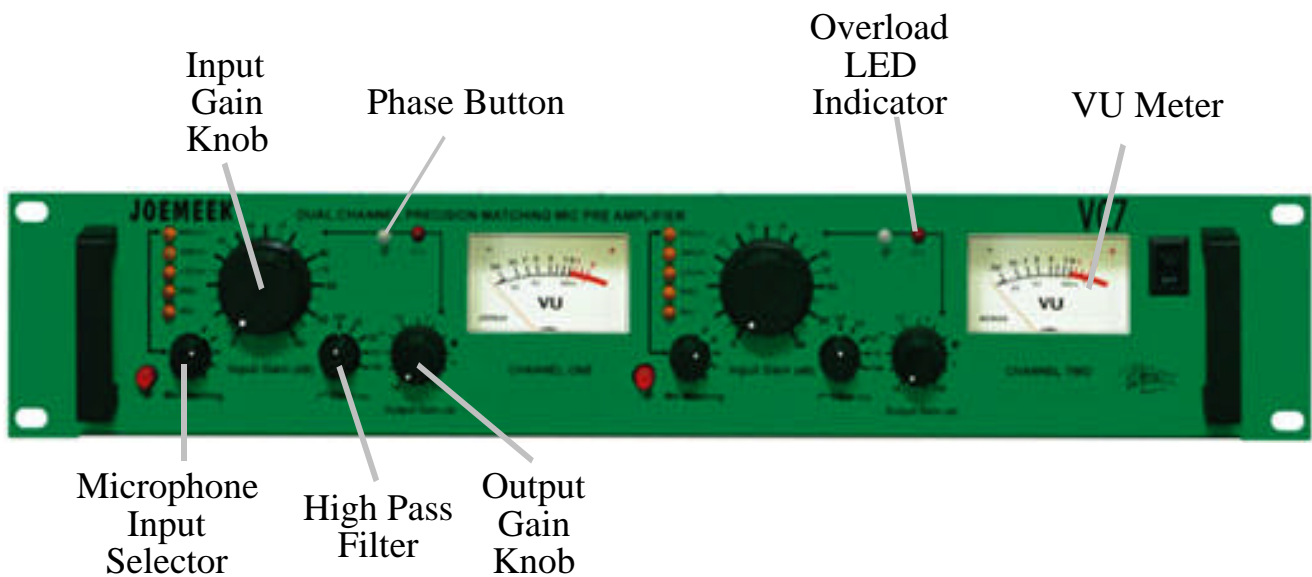
Set Soundscape to slave mode (press the **S** key on the computer keyboard).

Put Soundscape into play standby mode (click the play symbol on the transport bar or press the **↑** key on the computer keyboard).

When you are ready to record, start the VHS player. When Soundscape is running to the tape, i.e. when the big time code display is running, set Soundscape into record mode (click the red record icon in the transport bar or press **+** on the numeric keypad set on the computer keyboard)

The Foley artists performance is now recorded in sync with the tape in Soundscape.

## The Joe Meek VC7



The VC7 is a microphone preamplifier, well a pair of them in fact, and it has the electronic matrix circuitry that enables mid side microphone recording to be output as a stereo signal.

It also offers excellent microphone matching, so it can be matched with almost any microphone you want to plug into it.

Any microphone plugged into input sockets 1 and 2 in the sound booth are routed through to the VC7. There are two pairs of outputs routed through from there to the mixing desk. The first pair is routed to channels 1 and 2, and these are basically the signal straight through the unit, taking advantage of it's pre amps. The second pair is routed through to channels 3 and 4 on the mixing desk, and these outputs are those that have put the input signal through the mid side matrix to produce stereo output when a mid side recording setup is being used. If an A B recording setup is used, then the output from here is a mid side signal.

The following table shows the settings that should be used with different microphones:

'50ohm'	Classic microphones with 50 ohm impedance;
'200 ohm'	Use this position for all conventional 200 ohm microphones which do not require phantom power
'LOW Ph'	This setting is designed specifically for Rode, Neumann and Groove Tube microphones
'200Ph'	<b>Setting for most phantom powered capacitor microphones including AKG, Senheiser, Audio Technica and Joe Meek</b>
'600Ph'	Is for other capacitor microphones which have source impedances from 250 ohms up to 600 ohms

When using the Joe Meek Microphones, the setting to use is 200Ph, i.e. the setting for 200ohm microphones with phantom powering. It is important to match the setting here to the microphone in order to get the best signal to noise ratio. This setting is the one you will probably use most frequently. This adjusts the signal from the microphone to line level, which is the output level from the unit to the mixing desk.

A note of caution, don't ever use phantom powered settings, (those with Ph after the impedance number), with unbalanced microphones, it will damage the input transformer. The only unbalanced microphones we possess are the AKG D80 and the Senheiser Stereo Microphones. If you bring in your own microphones, please let Alan Brown inspect it before using it in the studio.

The VC7 has a phase button on each channel and this is used to invert the phase of the signal on the channel. Normally the indicator is green, but when used the phase light turns red, showing that the phase is being inverted.

The VC7 boasts a High Pass Filter that lets you set the slope point at 25Hz, 50Hz, 75Hz or 100Hz. This can be used if there is a low frequency content to the sound that you don't want to record. In the acoustically dead sound isolation booth, it is very unlikely that this will arise and you probably won't need to use this.

### Using the VC7

Make the sound in the sound booth and adjust the Input Gain Knob on the VC7 so that sound registers on the VU meter. It is good practice to set the input so that the average level is around 0 on the meter. It is OK for the level to go up to maximum on transient peaks, distortion should not occur, but make sure that the Output Gain Knob is set to 0 or below to avoid signal distortion at the output stage.

## The Compressor

The output from the mixing desk is routed through a compressor before it goes to the recording devices. The compressor is used to compress the dynamic range of the recorded signal so that you can smooth out a performance level wise. For example, if you have a narration where the narrators voice level is not constant, the compressor will help to make a recording that compensates for this.

The compressor used in studio three the Behringer Autocom Pro MDX 1400 is.

This compressor consists of three sections, the Expander/Gate, the Compressor/Limiter and the Enhancer.

### The Expander/Gate section.

An expander is used to increase dynamic range. It works by reducing the level below a given threshold level. The reason you might want to do this is if you have a noisy background to the sound you are recording that you want to reduce. For example if you were recording a guitar going through a chorus unit, the inherent noise in the chorus unit that you'd hear when the guitar isn't being played could be reduced in this way. In practice, the sound booth is acoustically dead, and so you are very unlikely to suffer this problem, so most of the time you won't need to use the expander.

The expander has two controls.

The **Threshold** control sets the level below which expansion takes place. It has a range of Off ~ +10dB. Most of the time this will be set to off, but if you do want to use it, set the threshold by gradually increasing the level until the offending noise is reduced enough but without affecting the level of the signal you do want to record.

The **Release** switch determines how fast the reduced level is restored to normal when the incoming signal raises above the threshold. (Switching the level without a fade would sound very distracting.) The settings are slow and fast, as a rule of thumb, fast would be used for percussive, short sound without much ambience and slow for sound with a longer decay.

Above the threshold control is the Threshold LED, the "+" LED lights up when when a signal above the threshold level is detected, and the "-" LED illuminates when expansion of the signal occurs.

### The Compressor/Limiter Section

The compressor works by reducing high levels of an incoming signal, but not affecting low levels. On this compressor there are two modes, "Hard Knee" and "Soft Knee". The former is quite severe in the way it initiates compression, the latter is more gentle and probably more suitable for most applications since it is more subtle and less noticeable.

The compressor has several controls grouped into sections. The first section is for setting the compression type and amount.

The **Interactive** switch is used to set the compressor mode between “Hard Knee” (more severe onset of compression) and “Soft Knee”, or Interactive, (more subtle compression for signals that are exceeding the threshold value but not going more than 10dB above it). The switch is in interactive mode when it is in the depressed position.

The **Threshold** control is used to set an absolute level above which compression takes place. It has a range of  $-40 \sim +20$  dB. The higher (more positive) you set this dial, the less you will compress the input signal. A good starting value for this would be +20dB.

The **Ratio** control sets the amount (severity) of compression expressed as a ratio. The ratio can be set between 1:1 (no compression) to  $\infty$ :1 (maximum compression, i.e. **Limiting**). An good starting value for this would be about 2.5:1.

The **Gain Reduction** set of LEDs above this section shows how much the gain of the signal is reduced by when compression occurs.

The **SC EXT** button is used to select an external signal as a controlling signal for the compression circuitry. Leave this switch in the inactive position, (not pressed in), since use of this function severs the connection of the input signal to the compression circuitry, effectively disabling the compressor.

The **SC MON** button is used to monitor a signal coming in to the unit via the side chain (a kind of “insert” function). It mutes the audio input and so this switch should be left in the inactive position (not pressed in).

The **SC Filter** switch is used to activate a high pass filter in the side chain path to limit the influence of low frequencies on the compression process. This switch will usually be left in the off position (not pressed in).

The second set of Compressor controls is used to set the rate at which the compression process responds to the input signal.

The **AUTO** switch is used to set the unit into manual or automatic response mode. Best results are obtained when this is in the depressed position and Auto is selected. In this mode the program material controls the attack and release rates to produce unobtrusive compression. If the switch is in the out position, then the following two controls can be set manually, otherwise they are overridden.

The **Attack** control sets the rate at which the compressor responds to the input signal when it exceeds the threshold. It has a range of  $0.1 \sim 200$  MilliSeconds.

The **Release** control sets the rate at which the compressor returns to its non-compressing state once the signal level falls below the threshold. It has a range of  $0.05 \sim 4$  Seconds.

The third set of Compressor controls sets the output level of the processed signal.

The **Output** control allows the output signal to be increased or decreased to compensate for the overall level change that is incurred by the signal processing. It has a range of  $-20 \sim +20$  dB. Usually you will want to have this control above 0dB to compensate for the reduced levels produced by compression.

The **I/O Meter** switch is used to switch the level LEDs above the output control. When the switch is pressed in, the input level is shown, when it is in the out position, the output level is shown. Usually it is more useful to see the output level than the input, so leave this switch in the out position.

The **In/Out** switch is used to enable/disable the dynamics processing circuitry, so if you don't want to compress your signal, have the switch in the OUT position, if you do want compression of your signal, have it in the IN position.

## The Dynamic Enhancer Section

This section of the unit is used to compensate potential loss of high frequencies that may arise in the compression process.

The **Process** control is used to set the amount of high frequency enhancement that occurs when compression is taking place. It has a range of Off  $\sim$  6, the higher the value, the more enhancement is applied.

Use this control with caution when recording voice overs, too high a setting may cause unnatural sibilance of the voice, especially for a deep voiced person, when high level low frequency content may activate compression and thus add extra high frequencies where they are not needed. It would be best not to use this control under these circumstances.

## Setting Everything Up

Set the controls of the expander/limiter as follows:

**Threshold:**        **70dB**

Whilst playing the source signal into the compressor, rotate the control clockwise until any background noise disappears. If you have no, or negligible, background noise, then leave this setting very low or off.

Set the controls on the compressor/limiter as follows:

**Threshold:**        **+20dBu**  
**Ratio:**            **2.5:1**  
**Auto:**             **In**  
**Output:**          **0dB**  
**In/Out:**          **In**

Whilst playing the source signal into the compressor, rotate the threshold dial anti-clockwise until a suitable amount of gain reduction is shown on the gain reduction meter. This will result in a drop in the level of the output, to compensate for this, turn the output dial clockwise until the output level is back up to a reasonable level.

Set the controls for the dynamic enhancer as follows:

**Process:            Off**

Rotate this control clockwise until the high frequencies of the output signal sound natural during compression, (when the gain reduction meter indicates compression).

This processed signal is ideal for recording to DAT and VHS, background noise will not be so apparent on quiet passages and loud sections will not distort.

Variations on the compressor/limiter settings will give different effects.

A low ratio with a very low threshold setting will give soft, musical processing of the general dynamics of the signal.

A higher ratio setting with low threshold settings give a relatively constant volume, (this is called levelling), useful for recording to unforgiving media like cassette recorders.

High threshold levels generally limit the overall level of the signal.

Ratios greater than 6:1 effectively stop the output level from going much over the threshold level, (as long as the output control is set to 0dB)



## Processing Mid Side Recordings In Soundscape

If you have opted to process your mid side recording in Soundscape, and this might be the preferred option since it gives you more control over the sound you produce, then this is how to do it...

First record your mid and side components into soundscape. Label them MID and SIDE. Duplicate the side recording using the copy between tracks tool:



Now use the phase invert tool to invert the phase of this copy:



Label it SIDE INVERTED.

Now mute the inverted copy of the side signal and activate the remaining two parts. Use the mixdown tool to mix these two parts together:



This is the left channel of the stereo pair. Label it LEFT.

Now mute the active side part and activate the side inverted part.

Perform a mixdown of this signal and the mid signal.

This is the right channel of the stereo pair. Label it RIGHT.

The advantage of doing this in Soundscape is that you can change the balance between the mid and side components to change the stereo effect.

Changing the mid signal volume changes the overall volume of the resulting sound.

Changing the volume of the side signal will change the width of the stereo image.

You also have the option of using your recording as a perfect mono sound if you simply use only the mid part!