JOEMEEK STEREO COMPRESSOR SC3

Handbook contents

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Thanks for buying the best compressor in the world.
If that causes a smile; just try it and listen!

WHY IT SOUNDS THE WAY IT DOES.

Conventional compressors used to be called 'levelling amplifiers'. They were designed originally to reduce the dynamic range of all program material so that it would record properly onto media that had limited range such as optical film and vinyl disc. As an engineering tool the compressor had to work in as linear a manner as possible and to be as unobtrusive as possible, so the attack characteristic (that is the way the changes in volume take place) had to be smooth, and the release (how the gain recovered when the audio signal was removed) needed to be long.

While experimenting with levelling amplifiers, engineers found that the application of compression had an effect on the perceived sound of music. The changes were subtle but definite. Equipment manufacturers over the years have tried to combine the requirements of engineers and produce compression devices which are both usable as levelling amplifiers and as effects units. This has been a completely wrong approach and has only served to create bad reputations for a number of products because, although they behave beautifully as engineering devices, they actually sound unmusical.

The JOEMEEK compressor is the first device commercially available to have been designed purely as an effects compressor. Its purpose is to change the way the ear perceives the sound; its action changes the clarity, balance and even rhythmic feel of music.

DYNAMICS

The human ear has a fantastic dynamic range; it hears and can interpret sounds from as low as a pin dropping, up to being next to a pneumatic drill. In fact, there are mechanisms in hearing which act as 'levelling amplifiers' and allow us to be able to hear and interpret this extreme range.

When very loud music is heard, there are two main mechanisms that allow us to make sense of the sounds. These are real biological compressors; the first one in 'software' affecting the way the brain interprets signals from the inner ear, the second is 'hardware' in the way the electrochemical impulses are passed from the inner ear. First there is a 'software compressor' in the brain that softens the effect of the loud sounds. This effect is fast acting and not long lasting. It acts over the whole frequency range and its effect is to soften the very loud peaks of noise or sound, if the sound is removed or stopped suddenly, then your 'ears' (actually your brain) recovers in just a couple of seconds to full sensitivity. Because we all live with the effect all our lives, it is rarely noticed.

MIMIC

The JOEMEEK compressor mimics this effect and instead of the listener's ear and brain doing the compression, the JM pulls down the sound at precisely the right moment and amount to fool the ear into thinking that the sound is louder than it actually is.
But doesn't any compressor do that? Actually no.

Although there are many compressors on the market that can be set to the correct time constants to mimic the effect, the actual gain reduction is never done properly. The problem is that the human ear is not an engineering device, it's non-linear in all respects. The psychoacoustic compression effect is like turning down the volume for an instant; but once there, the relative volumes of sound are NOT affected further.

Good Engineering Practice says that a compressor should work logarithmically; for a certain increase of volume, the output volume should rise proportionally less: That is, for a 2:1 compressor, an increase of volume of 10dB at the input should produce only 5dB increase at the output. a continuous process where the more you put in, the more it's pushed down.

The JOEMEEK compressor just doesn't work that way. As volume increases at the input, a point is reached where the compressor starts to work and the gain through the amplifier is reduced. If the input level keeps rising, gradually the gain reduction becomes LESS effective and the amplifier goes back to being a linear amplifier except with the volume turned down. And this is precisely how the human ear behaves! So the 'ear' is fooled into thinking that the JM compressed sound is louder than it really is; but without the strange psychoacoustic effect of 'deadness' that all other compressors suffer from.

ROCK CONCERTS

The second form of compression in the human ear is mostly mechanical. It involves certain parts of the inner ear being suppressed so that they do not respond to the large vibrations caused by loud noise. The effect is much slower to take effect and can take days to recover. This is the familiar 'deadening' effect that we all get in extreme cases when going to a rock concert. One of its effects is to change the way that we perceive different frequency ranges. Basically, the louder the sound, the more we hear of the HF and LF parts because our sensitivity to the mid ranges is reduced.

The JM compressor, because of its bending of the loudness contours has already fooled the ear into thinking that the sound is louder than it really is; the bonus is that the mid ranges seem to jump forward and the clarity improves because the mechanical changes that the brain thinks have taken place in the inner ear, have not happened.

With conventional compressors the 'fooling' effect is not so complete and the ear perceives something that is not quite right. Consequently, this additional clarity is missing and the whole effect is one of muddy deadness.Some historic compressors from the 1950s were used creatively by Joe Meek and other notable engineers of the past. They achieved some of these effects and the compressor models have become venerated for their 'sound'; the JOEMEEK compressor achieves more, and does it by design.

TIME

Of course the JoeMeek compound compression curves tell only part of the story. There are also the time constants of attack and release which have enormous effect on the musicality of the sound. The attack time produces audio 'punch' that again is a psychoacoustic effect, different attack times mimicking different levels of human biological compression.

Release time is even more important to maintaining illusion; the JM compressor uses a compound release circuit that reacts quickly to short bursts of volume, and less quickly to sustained volume, this helps to maintain the transparency of the sound.

A NEW DEPARTURE

The JOEMEEK SC3 has gone one step further than any other compressor with its control of release time; an additional control has been added to give finer and more precise control of transient compression. The startling results achieved with the SC3 make one wonder how anyone got along without it!

THE JOEMEEK SC3 MASTER COMPRESSOR: WHAT IT IS.

The SC3 is a revolutionary opto-electronic stereo compressor with precision balanced inputs and outputs designed to produce specific compression effects with absolute minimal noise and distortion in a professional studio environment. In addition, the compressor is fitted with highest quality and definition 24 bit digital input and output.

CIRCUIT ARRANGEMENTS: I/O

The circuits are arranged so that the compressor can be operated as a stereo analogue compressor, or an analogue to digital master compressor, or as a compressor working within a digital system.

- Analogue signal input is operational at all times.
- Digital input is switched on the front panel.
- The two types of inputs can be used individually, or mixed.
- The input gain control operates on both inputs and is callibrated for unity gain (digital and analogue mode).
- Both digital and analogue outputs are available at all times.

- The VU meter shows either compression attenuation, or audio level. This level is a mixed sum of left and right signals BEFORE the output gain control.
- The output gain control is callibrated so that at '0' the audio output corresponds to 0.775VRMS when the VU meter reads '0'.

ANALOGUE INPUTS.

Line inputs are 20Kohm floating balanced via XLR connectors maximum level is approx +28dBu.
5KV test isolation transformer inputs are available as an option with no change to audio performance.

INSERT.

An unbalanced port is provided so that additional analogue circuits may be inserted into the digital path.
The 'tip' is the buffered analogue output of the D/A converter, the 'ring' is the electronic switch input of the compressor (switchable from the front panel.)

ANALOGUE OUTPUTS.

Line outputs are XLR connectors 75 ohm source impedance, auto-centreing balanced. Maximum output +26dBu.

DIGITAL INPUTS AND OUTPUTS.

All digital connections DC isolated from analogue systems.
TOSLINK Optical inputs and outputs.
AES/EBU XLR input and output.
WORDCLOCK 44.1 or 48KHz output.

POWER

Power input is 230 or 115VAC at approx 12 watts.
Voltage selection is by rotating the fuseholder.
NOTE. Operation of the compressor on the wrong mains voltage WILL CAUSE TERMINAL DAMAGE.

CONTROLS AND INDICATORS.

- INPUT GAIN CONTROL is a rotary pot controlling the gain of both the analogue and the digital inputs to the compressor. The range of the control is from fully off, to a system gain of 16dB (with the output gain control at its callibration point).

- SLOPE is a FIVE position rotary switch which can be thought of as a ratio control, but actually changes the shape of the compression curve. It also sets the maximum amount of gain reduction in the system and has the effect of hardening or softening the compression.
  At position 1 the compression effect is small and the ratio low.
  At position 5, the compressor is approaching limiting; the ratio is about 8:1 at operating level, but less than this at low levels and during overload peaks.

- DIGITAL IN is a push button switch which enables the audio path from the D/A converter (incorporating the insert socket).
  A green LED indicates that the digital input is live.

- COMPRESSION is a rotary control setting the amount of drive to the compressor sidechain. The range is from off to maximum compression.

- IN/OUT is a push button switch to disable the compression sidechain for comparisons between compressed and uncompressed signals. A red LED indicates that the compression sidechain is active.

- VU is a push button that selects gain reduction or VU mode for the VU meter. A n LED warns that gain reduction is selected.

- ATTACK is a rotary control setting the rate at which the compressor starts to operate. At fully anticlockwise, the attack time for transient compression can be as low as 0.5milliseconds.
  The total range is about 6 milliseconds.

- TRANSIENT RELEASE is the most powerful control on the compressor. it sets the release time of transient compression. The range is about 10 milliseconds.When the control is at the 'normal' midway position, the effect is similar to the original JOEMEEK SC2 compressor.
  Varying this control provides a vast range of new compression effects. With the control towards minimum, transient signals cause the compression to operate as if the gain reduction was overshooting. This leads to an exaggerated 'compressed' sound. With the control towards maximum, the transient content of the signal has less and less effect on the overall compression and the result is of extreme smoothness.

- RELEASE is the normal release control which sets the time for any gain reduction to return to 'off' state.
  On the SC3 this control is very much less sensitive than on a conventional compressor.

- OUTPUT GAIN is a rotary control which sets both the analogue and the digital output level from the compressor.
  It is calibrated so that at the '0' mark, the noise and overload conditions are optimised. With the INPUT GAIN control set at its callibration mark as well, the through gain is unity.
  About 6dB of make-up gain is possible with the control.

- O/L LED. This LED is intentionally 'soft' in its action. The LED starts to glow slightly at normal operating levels and only becomes bright when the level approaches overload.

- ON LED is only there to let you know the unit is switched on; obvious really when the meter is illuminated as well!

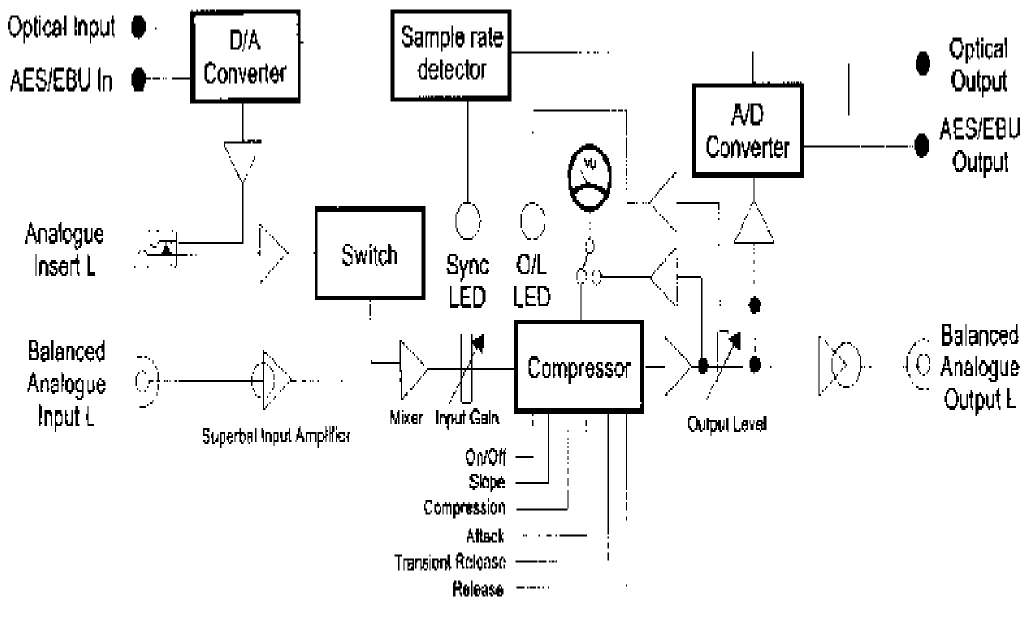
- SYNC LED shows when the internal digital oscillator is in 'master' mode.
  This is automatic when there is no sync digital input being received.

- On the rear panel there is a single push-button. This is the 44.1/48KHz sample frequency selector. Press the button in for 48KHz operation. It is ONLY FULLY OPERATIONAL WHEN THERE IS NO EXTERNAL DIGITAL INPUT.

When a digital input is detected, the digital system locks to the incoming frequency.

<u>CAUTION: IT IS STRONGLY ADVISED THAT THE SAMPLE SELECTOR SWITCH IS KEPT SWITCHED TO THE SAMPLERATE IN USE.</u> For example; when the SC3 is locked to 44.1KHz from an incoming digital signal, if the sample switch is set to 48KHz, following equipment could sense 48KHz instead of 44.1KHz. This can be important when recording to hard disk and with some CD-R systems.

DIGITAL PROTOCOL.
AES/EBU and SP-Dif formats have a selection of settings involving pre-emphasis and ident bits.
The SC3 is configured for 'consumer' protocol with no pre-emphasis as this is most generally recognised and universally operational in audio. The 'AES/EBU' input and output may be used directly with SP-Dif systems. The copy protect bit is stays set to 'copy OK'



Joemeek Stereo Compressor SC3 block diagram

DIGITAL AUDIO and USING DIGITAL SYSTEMS.

Digital inputs and outputs are becoming more and more common on professional sound equipment, but there is a serious lack of understanding of what digital is by most users. So a few moments going back to school!

Conventional audio is in the form of changing electrical voltages or currents which represent changes in pressure in the air representing sound.

Digital audio is a string of mathematical numbers which accurately represent those changing voltages. Each number is a sample. There are 44100 or 48000 samples taken each second. This 44.1K or 48K is called the sample rate. The digital number is called a 'word'.

OVERSAMPLING.

In both Analogue to Digital conversion and the other way round, it has become customary to use 'oversampling' techniques. This is to take the same sample many times, and produce a mean value for it and call that mean value the sample word.

WHAT IS 'CD QUALITY'

The standard 'type' of digital audio used on CDs is 44.1K, 16 bit.
The number of bits is the audio 'definition' that is achievable. More simply, it sets the dynamic range of the audio.
16 bit allows (theoretically) 65536 different levels of sound; and this was judged by the authorities as adequate for professional sound recording.

In practice, it means a dynamic range of about 85dB, distortion that is very difficult to hear, and a frequency range from nearDC up to about half the sample rate. This level of performance is achievable and repeatable; and is a good deal better than most analogue equipment!

PROFESSIONAL DIGITAL AUDIO.

But it's in the interests of the professional engineer to get the best possible sound quality, and to get something significantly better than 'CD' quality, it's necessary to use more bits. The JOEMEEK SC3 operates in 24 bit words and the oversampling is 128 times. This produces levels of distortion and noise that are only one fifth of those present in an ideal 'CD' signal. This quality and definition is useable if the digital signal is transferred to other equipment which can handle it. Otherwise the signal is degraded down to the level of the least good equipment it passes through. Although this sounds like a poor deal, it's still better than a complicated analogue path where signal quality suffers at every turn. Once in the digital domain, the quality is set at the 'worst' part of the chain, and stays there. Usually, that 'worst' part of the chain is the final CD recorder.

USING DIGITAL SYSTEMS.

It is considerably more difficult to interconnect digital systems than analogue systems. The reason is timing. For one digital box to talk to another digital box, they must both be working to the same digital 'clock'. This is so that the digital words are passed at the correct rate and they don't get mangled in the transfer. Fortunately, almost all digital audio 'receivers' have the ability

to 'lock' onto an incoming digital signal, and some can even adapt themselves to different sample rates (yes, the SC3 can). The problem remains that there has to be a master digital generator, and all the connected parts have to be 'slaved' from that one.
The SC3 has the capability of generating its own clock rate or adapting automatically to an incoming one. Provided that this clock idea is kept in mind, then interconnection is reasonably straightforward.

DIFFERENT SORTS OF DIGITAL?

There are 3 main types of digital interconnection used in professional audio.

1) AES/EBU This is an interconnection protocol based on standard audio XLR connectors. It's very solid and reliable and has the advantage of being able to use (shortish) standard microphone cables. The digital signals are transmitted like balanced audio so the theory is that interference is low.
In practise there's not much difference. The balanced digital signal appears across pins 2 and 3. It usually works best if the ground on pin 1 is connected at both ends.

2) SP-Dif This is a 'Sony' protocol based on the American 'phono' or RCA plug and socket beloved of Hi Fi equipment.
It's basically an unbalanced co-axial system that works well over short distances; say up to about 20 metres.
Although there is technically a difference in the digital word specification for SP-Dif from AES/EBU (SP-Dif incorporates an 'anti copy' bit which prevents second generation copying), in practice one will usually work with the other. The signal core of SP-Dif cable corresponds to pin 2 of the AES/EBU connector. The screen should be connected to both pins 1 and 3.

3) OPTICAL The connectors used for optical transmissions in audio are called TOSLINKS. The system is probably the best of all as one is not tempted to try to 'make do' with any old audio cable for digital connections.
The fibreoptic cable used is good for distances up to about 50 metres.

USING THE SC3 IN DIGITAL MODE.

During tests and field trials of the SC3 the most reliable and trouble free type of connection was OPTICAL. Although digital interconnection was always good by all types of connector, with some; particularly with the conversion from AES/EBU to SP-Dif, It was important to use good quality co-axial cable to get a 100% reliable link. YOU CANNOT JUST HANG IT ALL TOGETHER WITH HI FI WIRES!

Grounding is particularly important; when ground paths are wrong with analogue systems, the result is an obvious hum. With digital systems, it's not that easy; the system just doesn't work! (or in some cases, half works).
Use chunky cables and make sure that the grounds are well connected.

THE DIGITAL COMPRESSOR

When using the SC3 as a final mix compressor and a digital source (the main design use for the SC3), select the correct sample frequency with the switch on the back of the unit (out for 44.1K and in for 48KHz) then simply connect the digital output to the CD master computer or machine. Monitor via the master computer if possible. If the only useable output from the master computer is an analogue one I suggest it is ignored and that monitoring is done from the analogue outputs of the SC3. This is because the analogue ports from computer based equipment are invariably horrible. The quality of the digital signal will remain unscathed as it's already digital on the way in.

NOTE, The AES/EBU, the OPTICAL outputs, and the balanced analogue outputs can be used at the same time if required. There is no loss of quality in any respect.

THE DIGITAL INPUTS.

Unlike the outputs, the OPTICAL and the AES/EBU input connectors should only be used one at a time, otherwise the digital converter could become confused. Remember that the digital system in the SC3 will instantly adapt itself to any digital signal that is put on the input. That is; it will adopt the same timing and sample rate automatically it will ignore the sample rate switch. So when using the SC3 as a digital generator to other equipment, remember that connecting a digital input to the SC3 may alter the output digital conditions.

The Yellow LED on the front panel labelled 'sync' shows when the SC3 is providing the master sync to the outside world.
As soon as a digital signal appears on its input, the internal master clock switches off and the LED goes out.
This is a useful function as it tells when a viable digital signal is present.

THE INSERTS.

The insert facility is useful where access is needed to a signal in analogue form. The digital input converts to analogue and appears L and R on unbalanced jack sockets on the rear of the unit. The analogue signal is on the tip of the jack connection and any returning analogue signal appears on the 'ring' of the jack sockets.

CONVERSIONS.

When connecting an SP-Dif output to the AES/EBU input of the SC3, make sure that the screen of the co-axial cable is connected to both pins 1 and 3 of the XLR male plug. The inner conductor should be connected to pin 2.

LEVELS IN THE DIGITAL SYSTEM.

As the SC3 is a compressor designed to form the last part in the recording chain, the callibration point on the input gain control is set so that '0'dB on the VU meter corresponds to 10dB below digital clip level. This setting is a compromise between the 'under read' that occurs with a VU meter, and allowance for transient overloads.

The VU meter is a useful and simple indicator of overall levels but during mastering it is essential to check for any momentary overloads which could be overlooked by this metering.

The overload light on the front panel corresponds to about 8dB below clip.

When the output gain control is set to its callibration point, the digital gain through the system is approx +3dB.
This allows for enough lattitude to adjust the exact output level with minimised system noise and no danger of analogue clipping.

The SC3 uses a 24 bit digital word in its encoding, absolute levels are less critical as the system dynamic range can be as high as 125 dB. This means that it is extremely unlikely that digital noise will be audible above incoming analogue noise.

TECHNICAL SPECIFICATION.

ANALOGUE SIGNAL PATH.

INPUTS.
XLR connectors. Floating electronic balanced inputs with option of current mode transformer isolation. Input impedance 20Kohms. DC isolation (transformers fitted) 5KV.
Insert inputs unbalanced 1/4 inch jack.10Kohm nominal -10dB.

OUTPUTS.
Insert outputs unbalanced nominal -10dB source impedance 400 ohms.
Main analogue outputs XLR connectors. Auto centreing electronic balanced. Source impedance 75 ohms.

AUDIO PERFORMANCE.

AMPLITUDE FREQUENCY RESPONSE.: +0 -1Db 7Hz to 30KHz
HARMONIC DISTORTION : less than 0.04% at all frequencies and levels except under compression where 2nd harmonic distortion is generated by the compression.
MAX. OUTPUT : +26dBu
NOISE : less than -84dBu (dynamic range exceeds 110dB)

DIGITAL SIGNAL PATH.INPUTS AND OUTPUTS.

AES/EBU and TOSLINK OPTICAL.
WordClock output on BNC connector.
(Wordclock output follows samplerate switch or incoming digital sync).

CONVERSION PERFORMANCE.
A/D 1 bit system producing 24 bit word accuracy in the 24 bit wordframe at 128 times oversampling.
D/A 24 bit word capacity. 128 times oversampling.
Analogue performance limited at high frequency by sample rate. At 48KHz, system is 1 dB down at 22.5KHz.
Dynamic range capability 125B.