DSP

DIGITAL SIGNAL PROCESSING

Neve

THE DIGITAL AGE IS HERE

Digital technology touches our lives at every turn

Today's children treat computers as commonplace in school and home, whilst digital control brings benefits to the modern kitchen in an increasing variety of appliances from washing machine to microwave oven. In audio, as in most technical fields, digital methods have become standard in control and measurement, providing accuracy and convenience that are today taken for granted.

Now the sound signal itself is handled in digital form and we are experiencing today the most dramatic rate of development in audio since the era of the first electrical recording, and the dawn of broadcasting. Digital audio is now the standard medium for high grade recording and the Compact Disc is revealing to the critical home listener a new standard of audio quality which challenges many present technical and production techniques in the studio.

The home video recorder, where audio quality used to be the poor relation, is now capable of a stereo sound quality similar to the best FM radio, whilst satellite transmission will bring broadcast digital audio directly to the home. Video production and post-production studios must manipulate audio with the same care as the video itself receives.

Within this atmosphere of exciting technical change, creative production methods have themselves developed and cross-fertilized. Whether in audio and video recording, broadcasting, live theatre or concerts, this has resulted in substantial benefits for all entertainment media. For example, multitrack recording, once the sole province of the music recording studio, is today commonplace in radio, TV and film. Sound in the live theatre is exploring new creative realms thanks to today's technology. All this puts pressure on audio facilities. Mixing consoles must offer versatility, flexibility and speed of operation, whilst providing fully transparent audio quality

Once audio is in digital form – encoded into a stream of numbers – its quality is 'sealed in' and the signal is very robust indeed as long as it remains digital. However, most processes in the sound studio involve transfers from one tape to another for track-bouncing, mix-down, post-production editing or changes of level, equalisation, echo and dynamics. Without a digital mixing console, the signal must be reconverted to analogue form, losing the sealed-in quality, even for the most minor changes. Such processes occur many times in the production of the average recording and inevitably there is a loss of audio information at every conversion. The result can be that the final CD or broadcast fails to realise its full musical quality potential. The answer lies in Digital Signal Processing.

Immaculate sound quality is only the beginning

The DSP is one of those rare developments so far-reaching that it benefits immediately both technical performance and creative application. Digital Signal Processing has opened up whole new realms of automation and system flexibility which were previously only dreams. The ability to rearrange the whole system at will, total memory and instant restoration on all controls, and a range of special effects based on time delay, excite the imagination of sound engineer and producer alike.

What will be the technical requirements of music, TV, radio and theatre production a decade from now? It is a bold man who dares to make that prediction, yet that is the task facing today's audio console purchaser. He must be sure not only of handling the requirements of today, but also those of the foreseeable future.

Neve and Digital Audio

Neve has been supplying Audio mixing consoles and systems to the world's Broadcast, Music recording and Film industries for over 20 years. Neve is a dynamic and growing company – the world's largest specialist manufacturer of sound mixing consoles, renowned for audio excellence and versatility. The company now leads the industry into the digital age.

Neve's product 'Firsts' include the most exciting highlights in the development of Professional Audio. Following on the heels of NECAM and microprocessor-driven analogue systems comes the ultimate – Digital Mixing.

During the 70's, when digital audio was already a mature professional transmission medium, Neve feasibility studies revealed the exciting possibilities of digital audio mixers. When digital recording burst upon the scene with its potential for total audio transparency, the way forward was clear. Only by eliminating intermediate A/D and D/A conversions in the studio could the full capability of digital recording be realised – and only a digital mixing system could achieve this.

The implementation of equalisation, fading, mixing and dynamics control in totally digital form has taken the Neve development team right to the frontiers of digital signal processing. The results speak for themselves. Here is the most exciting audio system development ever. The Neve DSP Console system, with its versatility and hardware and software expandability will still be up to date and earning its living in the next decade.

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DREAMS FULFILLED

DSP SAVES YOUR TIME . . .

Every detail of set-up, EQ, dynamics, system routing indeed the setting of every control can be stored for instant recall *and reset*. At the stroke of a key, previous settings are restored and session set-up delays are eliminated.

DSP PUTS YOU IN COMMAND

How often have you wished that the console system was different for that special session? With DSP you can reconfigure the signal path at will, at the touch of a button. Fader layout may also be switched to suit your particular way of working by arranging that the controls most important to you fall most readily to hand.

DSP LIBERATES YOUR CREATIVITY

The unique processing power, with new features such as assignable time delay, in combination with the console's flexible mixing capability opens up whole new areas of innovation.

Memory and mixing automation operate on all main controls – EQ, dynamics, effects, cues etc in addition to faders, liberating the balance engineer to concentrate on musical creativity. All mix data is stored on floppy disk.

DSP TOTALLY TRANSPARENT

Digital processing means that DSP will only alter the sound if you wish it to – Direct Digital Connection eliminates any chance of deterioration.

DSP ALWAYS NEW

DSP is a console that you can 'build for yourself' – so flexible is the assignment system – so you can always keep up to date with the latest ideas. More channels and more processing power can be added without change to the control console. This system is never out-of-date – the best possible investment for the future.

DSP EASY TO INSTALL

The input units can be up to 1km away from the processing system, coupled by slim fibre-optic links that are immune from RF or EM interference. Each such link carries 16 audio channels plus 4 communication channels, so the number of lines to be installed is greatly reduced.

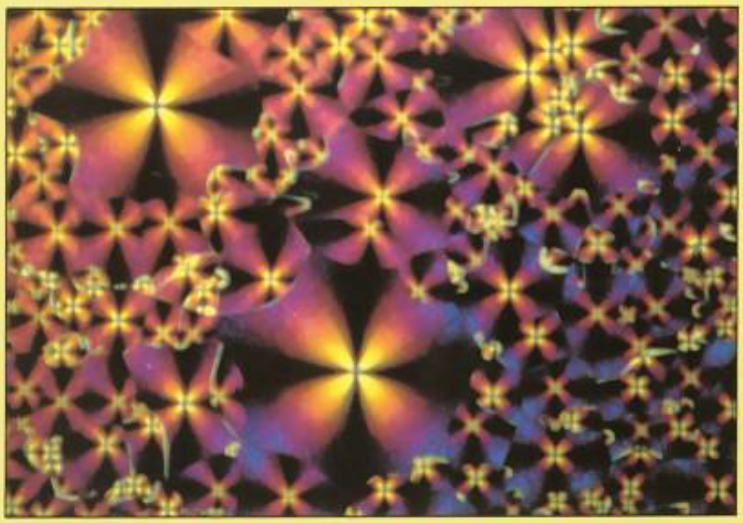
DSP FUN AS WELL

DSP is a user-friendly console, with a clear attractive pictorial display guiding the operator. 'Crazy' ideas are as easy to set up as routine systems – DSP will respond! This console lets you try out those wild ideas whenever you care to – the fulfilment of every studio client's dreams.

Neve

SHEER SOUND QUALITY	All-digital processing eliminates unnecessary A/D and D/A conversions, ensuring signal purity. Bit slice processing provides massive internal dynamic range (over 190dB in the mix bus) for perfect transients.
CONVENIENT IN USE	Your regular favourite configurations are available instantly on switch on. Creating specials is made easy by 'copy' and 'together' procedures enabling many channels to be set up together. All your special ideas can be stored for future use.
THE COMFORTABLE CONSOLE	DSP is human sized. All controls are within easy reach, with the vital ones nearest. There is plenty of room for hand and fingers to grasp the well-spaced knobs.
TRUE SOUND PERSPECTIVE	Every channel and group has available a variable time delay feature, enabling true perspective to be achieved by correcting for microphone position. Exciting special effects can also be readily achieved. Precision ganging of stereo pairs guarantees image position, with full sound stage width control available.
YOUR EARS THE JUDGE	The final arbiter must be the critical human ear. Digital routing flexibility provides the most comprehensive monitoring facilities. A separate monitor section – with full processing facilities – or in-line monitoring (or both) is available for maximum flexibility in multitrack operation. The convenience of assignable controls gives the operator the ideal centre stereo seat at all times, whilst the compact control console has minimal effect on critical control room acoustics.
TRIPLE GROUPING FREEDOM	Three methods of signal grouping provide additional versatility in complex mixing situations. True mixing, where sub-groups may feed others without noise generation. 'NECAM' style, ganging the motorized faders so that they move together. 'VCA' style, using the remote control feature of DSP to provide control from an assigned submaster.
THE AMBIDEXTROUS CONSOLE	At last the left-handed engineer is totally comfortable. Controls can be assigned as the operator wishes with convenience for either hand.
PERSONALISED FOR PRODUCERS	Special routing and configurations are easy to achieve and can be stored on a "Take away" floppy disk which becomes the Engineer/Producer's personal version for future use.
SPACE SAVING	The control console need only be as big as you want it to be. A 56-channel console is accommodated within an arm's span.
ANALOGUE-COMPATIBLE	DSP does not make the rest of your studio obsolete. Provision is made for analogue interfaces, not only at inputs and outputs but also in the form of insertion points freely assignable throughout the system, eliminating the conventional patch field.
KEEPING TRACK	DSP has an electronic 'scribble' – an alphanumeric display associated with each fader and control panel shows which source is under control. Re-assigning that source to another fader will move the display with it. These handy mimics extend to the studio mic racks to speed plugging up and of course are stored on the floppy disc for instant recall.
PULLS IN THE CUSTOMERS	Demand is growing for the transparent quality that only digital can confer, whilst the special effects and control automation are demanded by more and more studio clients.
TOTAL SYSTEM CAPABILITY	The vast processing power of DSP is not limited to audio. Versatile logic provides control of tape machines, studio communications, external effects units etc. making the DSP console the central command unit for the whole studio.

THE MANY FACETED CONSOLE



Few studios today can afford to specialise in one branch of recording alone. Viability demands adaptability, with studios serving both the Music Industry *and* Video or Film or Broadcasting.

Such flexibility makes unique demands on studio equipment – ideally calling for several kinds of control console as well as additional recording facilities.

The NEVE DSP is many consoles in one. Whatever the application or function required, DSP can be configured by the operator to achieve it. What's more, it can be reconfigured to another mode in moments. The DSP-equipped studio can switch from Film Dubbing to Music Recording without fuss and in less time than a musician takes to tune up.

To illustrate this flexibility the following descriptive account of DSP features is related to the specific activities of Music Recording, Broadcasting and Film. All the features are of course available in any application.

MUSIC RECORDING

Music recording today demands innovation – in sound and in techniques.

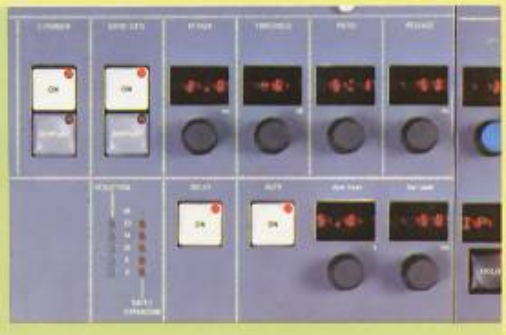
The DSP is the system which allows your innovative genius to flourish.

In addition to the usual Dynamics, EQ and Filter functions, an exciting new feature is available: Variable signal delay.

This yields many new possibilities: Equalising the arrival time of signals from microphones at different distances from the same source, retiming synthesizer tracks, or in combination with mixing and EQ, to create phasing and reverb or for that matter, effects which have yet to be named.

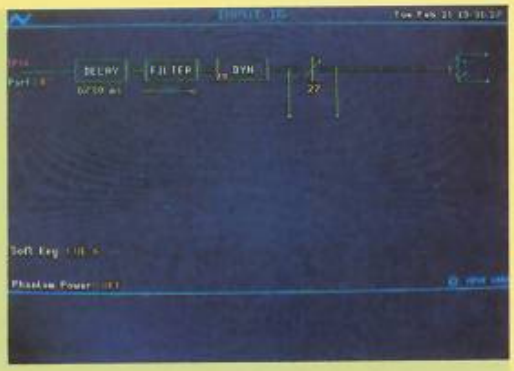
Up to 2½ seconds of delay is available per 24-channel processing unit which can be spread as required in any or all signal paths.

The DSP Dynamics processor is for correction and creation. Correction because even in the digital age there will be leakage between microphones in the studio, and artists may occasionally go 'Over the top' before the fader can be reached. But dynamics control is also a creative tool enabling sounds to be enhanced in a host of ways, which is why the DSP allows the facility to be assigned to all main signal paths.



The four main functions are independently switchable and may be set up to the desired parameters before bringing into circuit.

The DSP processor provides a signal gate function with wide threshold range, which thanks to the availability of signal delay, can anticipate the signal by opening immediately before the signal arrives. A flexible expander is also available, with variable slope and threshold, associated with which is an expansion display



bargraph. A similar gain reduction bargraph is provided for the limit and compress functions, which of course have variable slopes, thresholds, attack and release times, including release times that are programme dependent. EQ can be inserted into the side chain for vocal stressing or 'de-essing' and selectable signal path delay eliminates overshoot.

The VDU of the DSP shows clearly the position of any processor and its arrangement in the signal path in simple schematic form.

Dynamics side chains may be linked in any arrangement, for stereo operation, or driven from any external input to give gating or ducking effects. As may be expected in a digital system, all parameters are precisely defined and repeatable, and may be memorized for instant re-set.

Great care has been taken to optimise the signal level sampling method as well as the ranges of slope and time for the most responsive and musical effect. This Dynamics unit is a worthy companion to the famous NEVE 'FSE' equaliser.

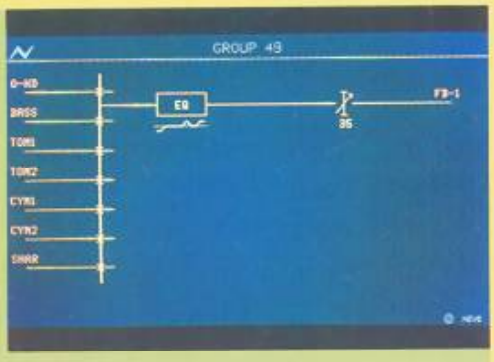
For Music recording, the DSP provides ample processing functions, so that each programme path, whether input or output may be provided with appropriate controls. The signal routing system is just as flexible, enabling the operator to set up 'nests' of sub-groups, and to achieve any desired track layout without cross patching.

In addition to these flexible processing functions the controls themselves are assignable, so that faders for instance can be organised in musical order regardless of where the microphones were plugged in, whilst other important controls can be placed near at hand.

Assignable multi-function controls are provided for each fader and can be switched to a variety of different functions at will so that, for example, cue sends for all channels can be displayed across the console.

Whilst track laying, a trial mix may be set up within the monitor section or using the main faders and processing functions as desired. Instant transfer from Track to Mix modes is provided at the touch of the Fader Flip button, with all controls set ready for the final mix.

Similarly, for fast overdubbing, one or more channels can operate as full recording inputs, routed directly or via any mix to the multi-track machine, whilst the remaining inputs provide the monitor and cue mixes of the existing tracks, together with the new contribution if required.



The DSP also manages the tape machine, setting the appropriate tracks into record as required, whether in simple overdub or 'Punch-in'.

During Mixdown, the great flexibility of the DSP is most evident. Sub-grouping, by true mixing (without the noise penalty of analogue mixing systems), by fader ganging (NECAM fashion) or VCA-style (but with no distortion penalty) are available in virtually any combination. Amongst the exciting new options open to the user is the use of signal delay within sub-mixing to produce effects equivalent to phasing, flanging and many others.

Of course, the DSP NECAM SMPTE time code automation option operates on all main controls and offers the easiest and fastest automated mixes yet.

TV AND RADIO PRODUCTION

Live shows need the DSP's instant flexibility.

Balancing a live TV show requires steady nerves and well-organised equipment. Frequently more than 100 microphones may be deployed in the studio, though it is rare for more than 20 to be live at once.

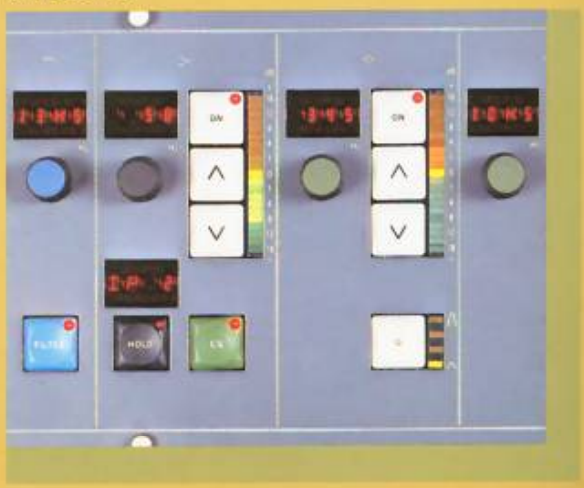
The DSP easily handles the necessary number of channels, and the option of two remotely switched mic inputs per channel gives further scope. Where successive scenes use different groups of microphones the DSP enables the operator to bring these under control centrally, with the appropriate preset EQ etc available in every case. As many as 100 preset mix patterns of this type may be stored on floppy disk for instant reset of the console between scenes.

The Grand Finale, with all available channels in use is equally easily handled, with preset balance stored for each microphone group, whilst the available faders are deployed as sub-masters or on critical channels. Again the fader arrangement will be as the operator desires.

Multitrack recording facilities are also available, with simultaneous mixdown controlled on an assigned fader bank or on multi-function rotary controls as required.

Of special value for broadcast use is the DSP equaliser, which provides the classic four function format with shelving bass and treble curves and two bell-shaped mid-frequency controls. All bands are tuneable over wide ranges and the mid-frequency controls have variable Q for versatile correction and effects. Numeric displays reveal the chosen frequencies in each band, whilst cut and boost are displayed on simple colour contrasting, bargraph displays, which together give an instant outline picture of the EQ curve chosen.

Associated with the equaliser are tuneable high and low pass filter sections which may be deployed either with EQ or independently.



Stereo operation has never been easier. The precise ganging of input, subgroup and output paths, with two-channel processing operated from one set of controls, gives a compact working layout and exact stereo tracking in all parameters.

Presentation (continuity) applications where many 'Outside sources' are involved can require elaborate Multiway-working or Mix minus matrices, which are easily established, thanks to the DSP's versatile mixing capability. A related requirement in Outside Broadcast (remote) use is effective talkback to commentators and others. This is a specific feature of the DSP input racks, which provide communication channels in the ratio of 4 for every 16 inputs.

A unique feature of DSP, especially advantageous in OB work, is the simplification of the link between console and microphone locations – 20 audio lines replaced by one fibre optic cable, which may be up to 1 km in length and is totally immune from electromagnetic interference. Each input rack has input sockets for 32 mics with remote switching to the 16 inputs allowing even more flexibility.

FILM AND VIDEO

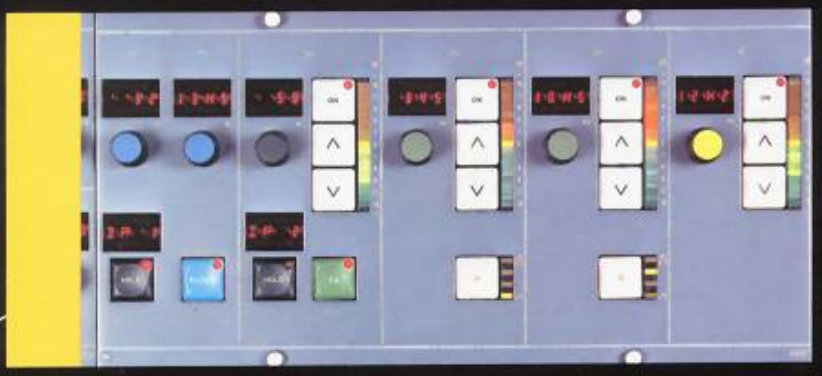
The attention to detail required by the exact and painstaking art of sound dubbing demands the DSP system. The generously spaced and uncluttered control panels give the balance engineer the most comfortable system, with all the processing power and accuracy that the task requires.

Special features to note are the combined numeric and graphic bargraph displays for EQ, the rapid acting dynamics processor and the precise automated pan-pots programmable to 3 or 5 channel characteristics.

Digital processing, allied to multitrack digital recording machines, provides a system that is free from 'generation loss'. The operator has the freedom to mix and sub-mix with as many intermediate recording stages as necessary without increase of noise or distortion.

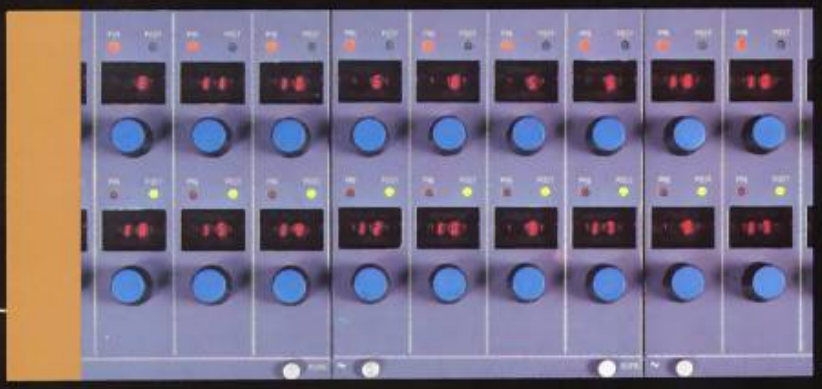
The adjustable signal delay allows the equivalent of 35 frames to be introduced into a single channel, providing a new tool for the adjustment of effects timing where the original recordings are not on independent machines. This extends the ability of the film studio to use multitrack as an alternative to sprocketed machines.

DSP NECAM post-production automation is as comfortable working in feet and frames as in SMPTE time code and provides a fast instinctive route to a well-polished sound track.

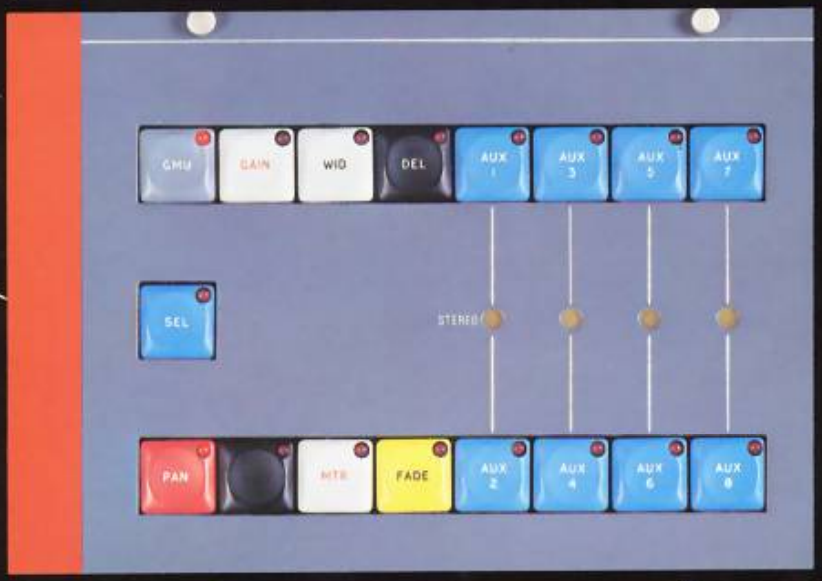


Wide-range filter and equaliser controls can be configured together or independently.

Floppy disk data store provides removable storage of all configurations, control settings and automated post production mixes.



Two multi-function assignable controls are associated with each fader to give multi-channel access to pan-pots, cues, effects sends etc.



Quick-access selector keys define operation of multi-function assignable controls.

An alternative monitor and communications panel.

THE DSP IN CLOSE UP

Integral high-resolution colour VDU provides full system status information.

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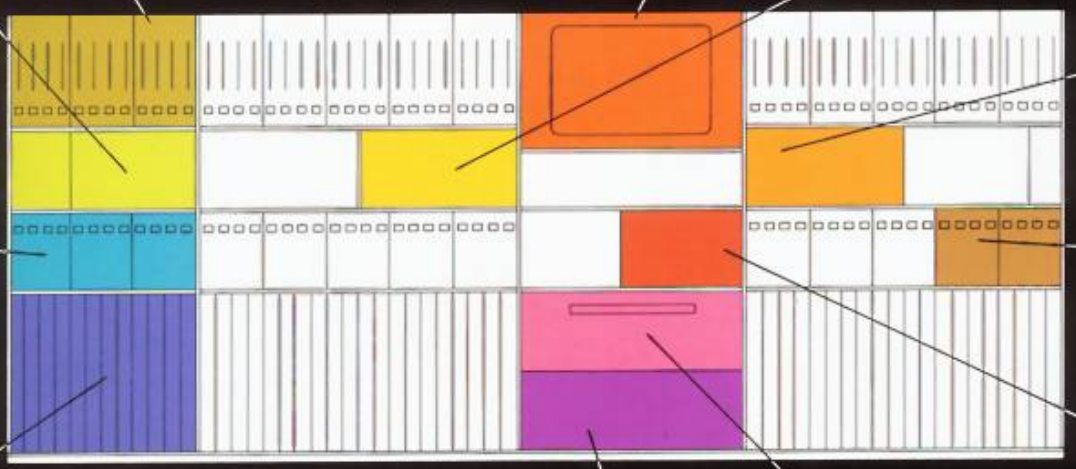
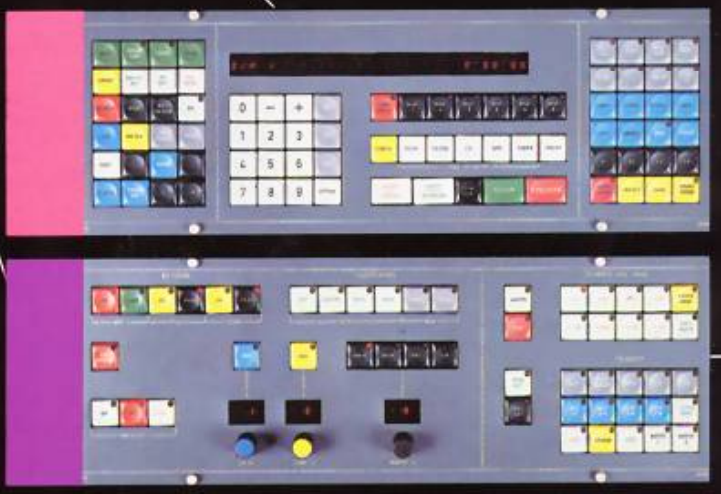


Versatile master routing/ assignment panel is the master controller for system configuration, store/recall and programmable functions.

Monitor and communications control panels are configured from a range of options to suit all requirements.



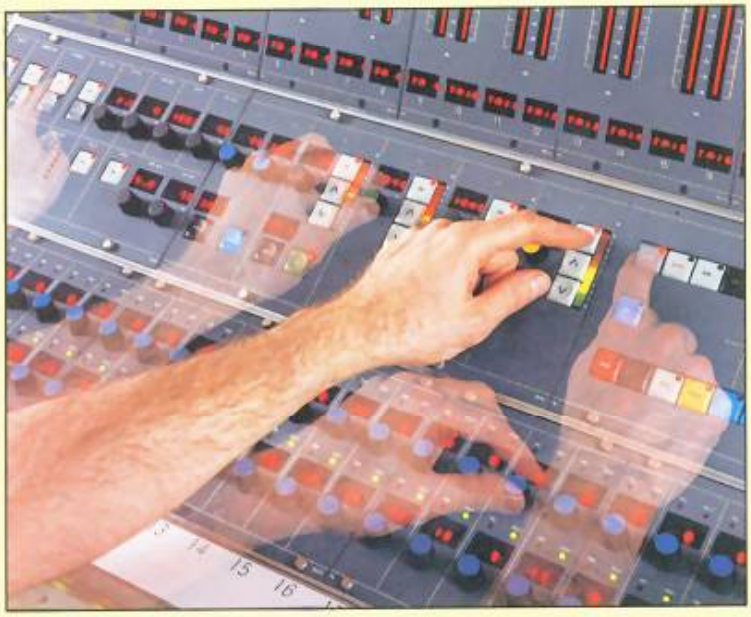
High resolution (200 segment) bargraph meters give accurate level measurement with versatile dynamic response.

Dynamics controls provide simultaneous limiter-compressor-expander-gate functions.

Facility panel provides access to communications levels and test oscillator. Auxiliary masters control levels to echo/effects/ cue sends.

Smooth light-action precision servo faders have integral function buttons, including source selector and programmable key.

WORKING WITH THE DSP



STARTING UP

Starting work with the DSP is quicker and easier. The start-up menu gives the choice of either setting the system to check its own state of health (diagnostics) or going straight into operation. If there's no time to run diagnostics at the start, don't worry – the comprehensive diagnostic software also runs during operation so that the main processing is fully checked out on a continuous basis.

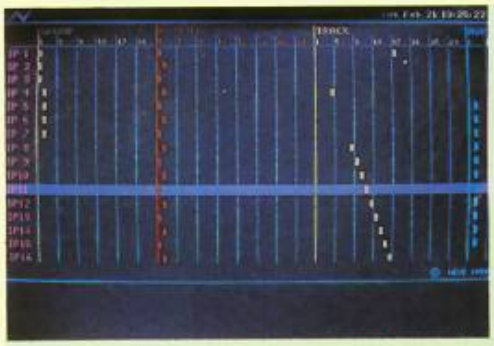
On selecting the operational mode, a range of options is presented on the colour VDU screen:

1. Continue exactly as the desk was set up when last switched off.
2. Continue as last time but with all controls zeroed ready for entirely new settings.
3. Basic configuration same as last time but with a new set of control settings recalled from floppy disk. This may be one of a series of static 'snapshots' or a mix from the optional time-code synchronised automation facility (DSP NECAM).
4. New system configurations may be selected from one of the four user-specified options embedded in console memory or something new selected from floppy disk. Most operators have their own favourites depending on the application, for example:

a) Tracklaying
b) Multitrack mixdown with overdub option
c) Direct stereo mix with clean feeds
d) 6 track music dialogue and effects for film or video
e) John's special dub with multiple EQ'd echoes and flange.

If previous control settings have been recorded on disk, then these may be restored to the desk by selecting the appropriate store from the menu.

Alternatively, a time-code synchronised post-production mix may be recalled and dynamic mixing continued from the last session. DSP NECAM operates on all processing controls including pan, echo, EQ etc, in addition to the faders, mutes and events.



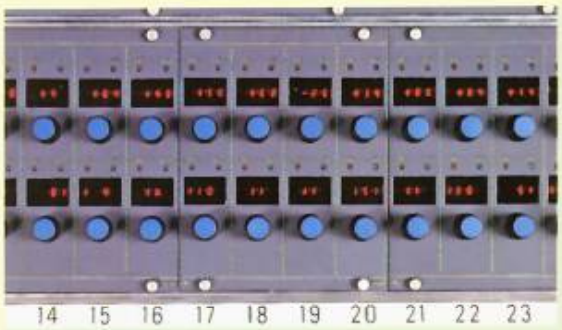
Whichever start up procedure is selected, real productive work begins at least an hour sooner with the DSP than with the usual analogue system.

OPERATION

Routing from any inputs to any outputs may be checked on the colour VDU and changed if necessary on the assignment panel by entering, for example, *"Input* 32, *Route* to *Group* 13, *Execute.* If preferred, input 32 and group 13 can be called up from their fader position simply by pressing the relevant access buttons.

With the comprehensive monitoring system switched to the required output, the recording or broadcast begins. Sound balance is set in the usual way on the input and group faders. Any input or group channel control such as EQ or echo send is brought to hand by pressing Access on the relevant fader, instantly bringing the set of channel processing controls on to the central assignable facilities panel for immediate adjustment. There is no more groping amongst packed channel strips. Next to the access button, PFL, AFL (solo) and cut buttons are conveniently available. Sometimes it may be useful to have a channel-by-channel display of a particular control, showing and accessing, for example, all the pan pots simultaneously, and perhaps all the echo sends. The two multi-function controls above each fader provide this facility, enabling all channel and group settings of a particular function to be seen and adjusted as necessary.

In multitrack working, the multi-function controls also provide convenient in-line level controls for multitrack sends (groups) and returns (monitor mix). Processing such as EQ and dynamics can be configured in any order and can be switched into multitrack (monitor) returns without sacrificing it in the corresponding input channel. Monitor mix settings can be transferred instantly to main faders for a quick start to multitrack mixdown.

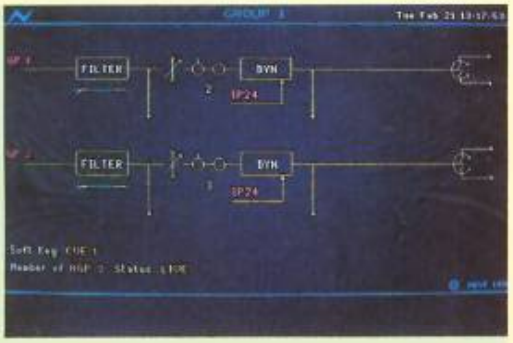


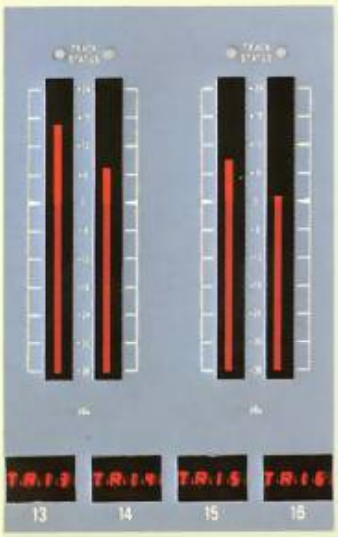
Many signals, whether from mic, tape or line, arrive at the console in stereo form. These are conveniently handled simply by defining two inputs and relevant mix groups as a stereo path so that their respective faders, equalisers, dynamics units etc are all ganged and work from a single set of controls, including full stereo image width and offset effects. At any time, pre-assigned routing and control setting snapshots which have been previously memorized at rehearsal can be instantly recalled to the console either in sequence or at random, greatly easing pressure in critical sessions.

METERS AND MONITORING

The all-important signal levels are displayed on a highly sophisticated metering system handling both analogue and digital signals. The bargraph meters give a clear PPM indication with their smooth inertia-free movement over the very wide level range from –36dBu to +24dBu. Digital recording benefits from the absolute peak and peak hold characteristics which give true "zero rise time" reading of the fastest transients. For those applications where the traditional averaging properties of the VU meter are needed, the bargraphs can be switched to this characteristic too. For a comprehensive level indication, any two of the four meter characteristics can be displayed simultaneously using the "bright up" ability of the plasma bargraph display.

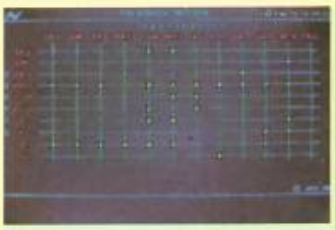
The versatile monitoring allows simultaneous or separate feeds to up to 8 pairs of loudspeakers from all main and auxiliary outputs and external sources whether analogue or digital. A monitor link to the system routing panel provides a 'listen' facility at virtually any point in the system, and a pair of meters follow monitor or access as required, giving an immediate reminder of the most important levels.





EFFECTS AND INSERTIONS

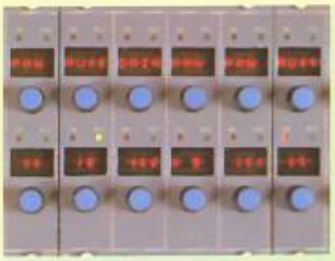
Even with all its inherent versatility and time-delay effects, the DSP does not pretend to replace totally the established known and loved effects and echo units – every producer and engineer has a favourite depending on the occasion. Fully assignable analogue or digital insertion points can be moved around anywhere in the system, whilst the echo and effects send and return architecture has great flexibility. Special software can provide an effects mixture control on every channel which keeps total signal level constant whilst varying the proportion of signal fed via the echo or effects device. Once that echo proportion is set, it can be held constant despite group fader level changes. The 12 independently controlled auxiliary sends available on every channel, group and track return are switchable as effects sends or cue/foldback/PA feeds. All can be directed to the comprehensive output routing switcher for maximum versatility. System intelligence differentiates automatically between cue and effects sends to ensure correct communication routing and solo/muting logic. Effects returns are likewise identified separately from other input paths for isolation from cut solo.



PRODUCER, AUDIENCE AND SUPERSTARS

The demanding task of the sound engineer does not stop just with the perfect mix. Producers, performers and studio audiences must all be looked after with foldback sends, cue lights, PA and talkback communications. Whatever the situation, live mix, initial tracklay, over-dubbing on a spare track or doing a critical punch-in, cue/foldback sends and talkback give the performer the most natural conditions possible, with the minimum distraction at the punch-in point. Two-way talkback between producer and performer is free and open except when the intelligent security logic sees the tape machine go into record and mikes are switched through to the relevant tracks.

The sensitive task of providing each performer with the 'right' cue mix is eased by the 12 auxiliary sends, which may be paired off as stereo cues as required. A fast, easy starting point for each cue mix is obtained just by taking the mix from main or monitor mixdown on to the cue sends.



Talkback and foldback sends, whether to loudspeakers or headphones, give the right signal at the right level and only mute if the system detects a 'live' studio mike under record or broadcast conditions. This intelligent security system has full knowledge of system routing and fader settings, so that its powerful interlocks ensure no embarrassing accidental broadcasts of foldback, talkback or feedback and look after the ears of the superstars by never feeding nasty noises to cans.

The critical job of PA sends to a studio audience has sometimes involved uncomfortable compromise. The DSP gives maximum feedback-free level using a number of features.

Pre-fade PA sends are automatically muted if the channel fader is closed, minimising number of 'open' mikes. The comprehensive DSP EQ dynamics and delay may be inserted in all or part of the PA chain to deal with difficult acoustic conditions. Multi-function controls on the assignable console give immediate fast access to any particularly critical PA send. For very critical theatre sound applications a small 'slave' console linked by slim optical fibre to the main unit can be operated in the auditorium in optimum listening conditions.

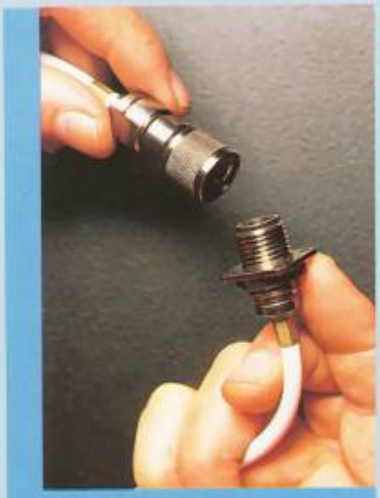
TECHNOLOGY IN HARNESS

This description of DSP operation can but hint at the enormous power of its intelligence and memory provided to serve the user. The clear alphanumeric displays of path names carried right through to mic sockets in the studio, the easily followed coloured system "maps" on the VDU, and the precise control level displays ensure that the operator is always in total command with full information at his fingertips.

The ultimate test of any new product is the degree to which it can harmonise with the skills of operator, performer and producer to create something greater than the sum of the parts. The DSP is proving to be a powerful new medium for creativity in the studio. Let the results speak for themselves.

THE PHYSICAL CONFIGURATION

The NEVE DSP is much more than a console, it is a digital studio system. There are three main sections.



INPUT RACKS

Accepting up to 16 microphone or line inputs each, these units provide high quality, low noise analogue amplification of microphone signals, followed by conversion to digital signals which are multiplexed onto a single fibre optic link for transmission to the main processor rack.

Available from the same unit are four return channels for cue or talkback. These racks are mains powered and may be distributed within the studio so as to provide convenient access for microphone or headphone connections. The unique properties of the fibre optic link not only simplify studio wiring, but provide total immunity from RF and other electro-magnetic interference.

The microphone inputs are balanced and transformerless and are provided with two input sockets each. The choice of input socket, and of amplifier gain are digitally controlled from the console itself.

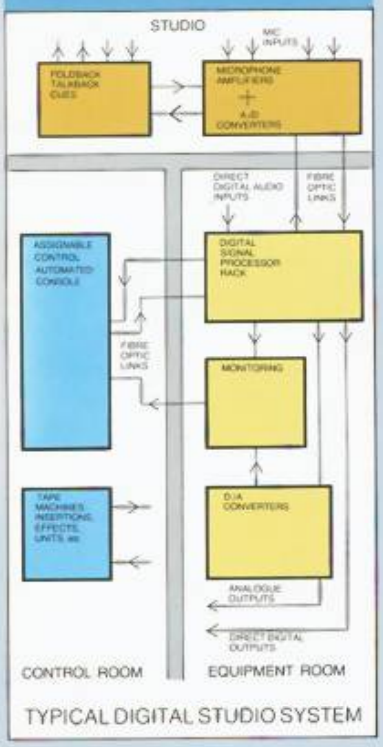
DSP PROCESSOR RACKS

Usually contained within three 36U (6ft) cabinets, these are the heart of the system and are entirely remote controlled from the console by digital instructions, again linked by optical fibres.

Direct digital inputs are available for replay or mixdown of digitally recorded signals, whilst similar outputs are available for master recording. These signals conform to the AES/EBU interface standard, with a normal sampling rate of 48kHz.

Analogue outputs are also available for monitoring and to feed analogue tape machines and lines.

The optical fibre links from the processing racks to the control console and input racks may be up to 1km in length, thus imposing virtually no limitation on studio layout.



TYPICAL DIGITAL STUDIO SYSTEM

CONTROL CONSOLE

Neither the audio signals nor their digital equivalents pass through the control console itself. Control positions and settings are themselves digitally encoded and coupled to the processor racks by fibre optic cables.

The console is simply an array of controls with associated graphic displays. This yields two advantages – firstly, in giving freedom to design slimmer and more ergonomic structures, and secondly, in enabling full assignability of controls, thus reversing the trend towards giant consoles crowded with controls. With the DSP the console has regained human dimensions with comfortable reach and ample finger clearance.

Associated with each control is an alphanumeric display which indicates the setting level, frequency, boost, time etc. Further displays show the circuit under control (CH23, GP4, BASS, VIOL etc).

The colour VDU not only presents clear maps of signal path configurations but also provides the operator with running instructions in the form of Start-up and Test menus.

NEVE SUPPORT

When you invest in a NEVE DSP, you are not only buying the most comprehensive console ever made, you have available a first class world wide maintenance capability.

NEVE back-up helps you at three levels:

Firstly, comprehensive in-built diagnostic systems automatically verify the operational status of the system during operation and will indicate fault location to facilitate appropriate action, whilst side-stepping where possible to continue normal functions.

Secondly, NEVE's Service Engineers are experts at 'Telephone diagnosis' – interpreting the observed symptoms in the light of their experience, and able to despatch replacement items rapidly when needed.

Thirdly, NEVE Engineers can be on their way to your studio within hours, anywhere in the world.

DATA

Definitions

0dBu=0.775 volt irrespective of impedance

Noise is measured at 20°C, r.m.s. unweighted in the band 22Hz – 22kHz as defined in I.E.C. recommendation 268.1.

Gain is defined as 20 log Vout/Vin where Vout and Vin are measured at the device terminals.

Processing facilities

Equaliser

Four bands with individual in/out switches in addition to overall EQ in/out.

Low frequency band:
shelving frequency variable from 30Hz to 600Hz in 21 steps.

Low-mid band:
peaking frequency variable from 100Hz to 1200Hz in 21 steps.

High-mid band:
peaking frequency variable from 600Hz to 12kHz in 21 steps.

High frequency band:
shelving frequency variable from 1200Hz to 15kHz in 21 steps.

Maximum boost/cut available in each band is ± 18dB (21 steps).

Low-mid and high-mid bands peaking characteristics have variable Q selectable to four values (0.5, 1, 2, 5 measured at maximum boost). Relationship between Q-value and boost/cut is defined for optimum musical effect (Formant Spectrum Equalisation characteristic).

Filter

High pass:
–3dB point variable from 25Hz to 280Hz in 21 steps.

Low pass:
–3dB point variable from 2kHz to 15.7kHz in 21 steps.

Roll-off 12dB/octave (Butterworth type filters).

Dynamics

Gate attenuation 30dB

Expander slope variable from 1:1.25 to 1:16 in 9 steps.

Compressor slope variable from 1.25:1 to 20:1 in 10 steps.

Limiter slope 100:1.

Gate threshold variable from –48dBu to –10dBu in 2dB steps.

Expander threshold variable from –28dBu to +10dBu in 2dB steps.

Limiter threshold variable from 0dBu to +18dBu in 1dB steps.

Compressor threshold variable from –20dBu to +18dBu in 2dB steps.

Attack time variable from 20µs to 50ms in 10 steps.

Manual release time variable from 50ms to 5secs in 8 steps.

Auto release facility allows free selection of 2 independent release times for fast peaks and for mean signal level to give optimum subjective result.

Signal Delay:
200µs switchable to allow side-chain anticipation.

Time Delay

A total of 2.6 seconds of audio storage is allocated for each set of 24 channels of signal processing, available to be assigned freely by operator as first function in input or group signal path e.g. every processing channel can have average 108ms delay simultaneously. Control is in 20µs steps up to 400µs, 100µs steps up to 2.5ms and 1ms steps thereafter.

Performance figures

Mic/line analogue inputs –

– balanced, transformerless

Gain range:
–26dB to +80dB in 2dB steps ± 0.3dB.

Input impedance:
1k2 ohms ± 5% in parallel with 840pF over gain range +28dB to +80dB. 10k ohms ± 5% in parallel with 840pF over gain range –26dB to +26dB.

Common-mode rejection:
> 65dB over frequency range 40Hz – 15kHz.

Equivalent Input Noise:
< –125dBu for mic gains > 60dB (200 ohm source).

Input headroom:
> 30dB (analogue region) > 46dB (digital region). Figures are relative to 0dBu line-up level and are subject to the absolute maximum input handling capacity of +26dBu.

Crosstalk:
between input 1 and input 2 on a single balanced input module (input-referred).
> 150dB at 40Hz
> 140dB at 1kHz
> 110dB at 15kHz

Analogue outputs

– balanced, transformerless

Maximum output:
+24dBu into 200 ohm balanced load.

Stability:
unconditional.

Short-circuit tolerance duration:
indefinite.

Digital inputs and outputs

Differential biphase-mark serial transmission to RS422 standard.

Transmission format in accordance with AES/EBU recommended practice for linearly represented audio data. Each copper wire pair carries two channels at 48kHz sampling rate (44.1kHz to special order). Digital audio data occupies the 16 most significant bits (12 to 27) of the 32-bit word. Expansion provision is available for 20-bit interfacing. Special interfaces to digital tape machines of different formats available to order.

Peak digital signal level ($.7FFF_{16}$) equivalent to +24dBu r.m.s. on analogue I/O.

A/D and D/A system performance

Sampling rate:
48kHz (44.1kHz to special order).

Quantisation:
16 bits.

All inputs optimally dithered.

Signal: noise ratio (maximum signal level):
> 89dB Adjustable dynamic range window gives optimum performance with all analogue sources and destinations.

Frequency response:
20Hz – 20kHz: +0.2dB – 0.5dB.

Total harmonic distortion at +22dBu level, 20Hz – 20kHz: < 0.05%.

Anti-alias and anti-image filters phase-corrected. Variation from constant phase delay over 200Hz – 15kHz frequency range: ± 6µs.

Maximum phase error between two nominally identical paths over 20Hz – 10kHz frequency range: ± 5°.

Crosstalk between any two paths through entire signal chain (measured relative to hostile path).

40Hz: < –80dB
400Hz – 7kHz: < –90dB
15kHz: < –80dB

DSP

'The results speak for themselves. Here is the most exciting audio system development ever. The Neve DSP Console system, with its versatility and hardware and software expandability will still be up to date and earning its living in the next decade.'

Neve

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