A. INTRODUCTION

The EELA AUDIO S340 is an innovative approach to the design of a multipurpose broadcast control desk, based on the experience with our S440 design and a lot of user feedback on it.
We have found that the operator interface is very much a matter of personal preference, previous experience, the type of program and the application of the system for direct ON AIR or for production purposes.
Also interfacing to a large range of source equipment, like different types and makes of machines has to be very flexible, mainly because there is little or no standardisation in this area.
The answer we found for this great variety of configurations is the use of a MICROPROCESSOR in each channel for controlling the logic and the switching of the audiopaths. By means of programming switches the way of working can be altered by the user to adapt this to his needs or wishes.
The use of an individual processor per channel is mainly a matter of safety and ease of servicing. All communication between modules is in the "old fashioned" way by means of static DC & LOGIC busses.
All switches used in the logic system are keyboard type pushbuttons with LED indication and replacable legending, with the exception of the ON/OFF switch, which has a lamp as indicator for easy visibility.

The processor control allows for future updates of the logic functions and is easy for customising without changes in the hardware.
The architecture of the audio- and DC control busses provides for quite a large range of custom setups, which can be implemented at low cost by simply replacing the EPROM's. Several versions of the software will be available at a later date.

Two types of inputmodules are available now: for high level stereo lines and for mono sources being either microphones or line/telephone circuits, each with two independant inputs, dedicated logic control and interfacing to the source equipment.

Connections are made with XLR's and D25 multipole connectors for in- and outputs, with 1/4" jacks for the insert-points and returns outputs, and with D9 multipole connectors for channel remotes.

The master module houses all mix- and line amps and the motherboard underneath is used for distribution of all audio-, control- and power signals to the modules, the facilities modules in the toprack and the "outside world".

B. SYSTEM ARCHITECTURE

1. AUDIO MIXING BUSSES

The audio mixes are made with a differential balanced virtual ground summing system with high rejection of both RF and LF noise. The crosspoints are made with CMOS switches for low noise operation. For maximum signal to noise ratio the crosspoints are opened when the contributing channel is OFF. Control of the switches is via the channel-processor.

Available (stereo) mixes in the ON AIR setup are:

1. MAIN (AMIX), postfader outputs from all channels.
   A sum of all modules creates the MAIN outputsignal, used for transmission and/or recording.
2. BMIX, postfader outputs from all but the microphone channels assigned to STUDIO with the STUDIO mic-input selected.
   This mix is or can be used for creating the signals for loudspeaker foldback in the studio (N minus local microphones).
3. SEND (CMIX), switchable post fader output without levelcontrol, for free use.
4. AUX, postfader (stereo) output with levelcontrol.
5. PFL (PREFADER LISTENING), switchable prefader stereo output to be used for checking the signal for level and quality and cueing the source equipment. Several logic facilities are coupled to this function.
6. COM (COMMUNICATION) / LST (LISTEN), switchable prefader mono output in use for communication purposes, e.g. audible contact with studiomicrophones, telephonelines etc. Most designs use the PFL system for this, a separate system has proven to be more comfortable.
   Can also be used as LST (LISTEN), a switchable prefader mono output, being a second prefader listening system, that can be controlled externally from the desk and is of value for a presenter or producer to listen to sources, independant of the engineer.

These audio busses are located on the 50 pole ribboncable, connected to the mainboard in every module.

A total of 11 balanced outputs are available for:

1. MAIN LEFT and RIGHT (transformer balanced) for driving the transmitter or the lines centre.
2. MONO (transformer balanced) output.

Other mentioned outputs are electronically balanced:

3. RECORD LEFT and RIGHT as source for recording machines, utilising the MAIN signal via separate line- amplifiers.
4. BMIX LEFT and RIGHT.
5. CMIX LEFT and RIGHT.
6. AUX LEFT and RIGHT.

2. DC LOGIC & BUSSES

These busses are used for communication between modules on logic- and DC control level. The following signals are present:

1. PFLDC (PREFADER LISTEN DC), for controlling the monitoring and metering when a channel activates the PFL system.
2. PFLRST (PFL RESET), a reset bus for switching off former selections to the PFL bus when an other channel is activated. This function can be disabled to give a mixed PFL behaviour.
3. COMDC (COMMUNICATION DC), for controlling the intercom speaker and/or the presenters headphones when a channel uses the COM audio bus or when the station intercom system demands access to the desk.
4. MUTE ST, a signal active on opening microphonechannels assigned to STUDIO for controlling monitoring and signalisation in the STUDIO location.
5. MUTE CR, the same signal, but than activated by CONTROLROOM microphone channels.
6. CGH (COUGH), a logic input to microphonechannels assigned to STUDIO, for, when enabled, muting the channel and opening the COM audiopath for communication from studio to controlroom on initiative of the studio.
7. TMR (TIMER) , a continuous- and a pulsed signal on this bus is used for RUN and RESET of a machinetimer or studio stopwatch, active on opening stereochannels with the function enabled. This way a runtime indication of the last started machine is automatic available. The channels can be selected active both from preset switches and from the frontpanel.
8. DIM, activated by channel talkback functions (mono line module) for opening the talkback audio signalpath and lowering the listening level in the controlroom.
9. LIMIT, a bus connected to all channel VCA control circuits for simultaneous gaincontrol of all channels, either manual (MAIN MASTER FADER) or via a limiter/compressor module for overall limiting.
10. TX, a serial TRANSMIT output from the modules, for future use.
11. RX, a serial RECEIVE input to the modules, for future use.

These busses are also located on the 50 way ribboncable on the module mainboard, combined with the + and - 18 Volt audio supply .

3. HIGH LEVEL BUS

This buscable carries high level audiosignals used in several inputmodules.

1. RECL (RECORD LEFT), a balanced high level bus with the MAIN LEFT signal, coupled to the machine connectors in the stereomodules for recording the MAIN LEFT signal.
2. RECR (RECORD RIGHT), the same, but than with the MAIN RIGHT signal.
3. TBTAU (TALKBACK AUDIO UNBALANCED), unbalanced talkback audio from the desk for communication to the mono line channels in 2-wire or 4- wire use and the speakers and headphones in the studio. The signal can be derived from the internal desk microphone in the master module, a presenters microphone or the intercomsystem.
4. TBRAU, talkbackaudio from a producers desk for the same purposes.
5. CFMU (CLEAN FEED MONO UNBALANCED), unbalanced out of phase MAIN MONO for deriving N-1 mixes in the mono line channels to be used as returnsignal to the caller.

These busses are located on a 10 way ribboncable. Also the 48 Volts Phantom powering runs over this ribbon cable.

4. GROUNDING SYSTEM

In order to keep the signal as clean as possible, in spite of all the logic, LED's, lamps etc, we have used a diversified ground system in all modules.
The xxxG (reference ground for all mixamps) is a seperate bus for each audiomix to be returned to the corresponding mixamp.
The GND (analogue ground),is as power return ground for both audio, microprocessor and related circuits.All ground currents from opamps, LED's and switches are returned to this ground, while lamp- and relay power is switched between the audio powerrails, not to any ground bus.
The CH GND (chassis ground) is the frame of the console which has to be connected to the protective mains ground. Audio ground has to be connected to the chassisground preferably on one place only being the back of the mixertray on the seperate binding posts.

5. POWER SYSTEM

The main audio powerrails are + and - 18 Volts, decoupled in each module for isolation of faulty modules from the system and for improving LF crosstalk.
The + 5 Volts for the processor, the logic circuits, the LED's and the VCA fader reference are made with an individual stabiliser per module from the main audio powerrails.

C. DUAL MONO INPUTCHANNEL S341

This module is meant for control and processing of mono microphone- or line signals. The logic system and the audio routing are laid out for simultaneous use of up to two locations for the microphones, being the CONTROLROOM and one STUDIO with facilities for foldback control, communication and signalisation.
Each input A or B can be configured for one of the 2 locations.
When using any of the inputs as mono line input this wil not only affect the audio path but also by determining what kind of mono source is connected (hybrid or music line) this will also be taken in consideration by relevant logic controls. An electronically balanced return output per module is used for send to hybrids, 4-wire send or booth foldback carrying the main signal minus the signal of the selected input.

The function of the inputs being either MIC/MIC or LINE/MIC or LINE/LINE is determined by the kind of connectorpanel used at the back. This is specified at ordering the console.

1. <u>OPERATOR CONTROLS</u>

1.1 INPUT GAIN CONTROL

This is a pot with a wide range from 20 to 70 dB for the microphone inputs and a limited control over a +/- 10 dB range for the line inputs.
The inputstage is a sofisticated low noise dedicated differential balanced input circuit, which can be modified by adding a transformer if galvanic isolation is a must.

1.2 HIGH PASS FILTER

A continuous variable control for setting the frequency of the high pass filter with a range from < 20 Hz to 200 Hz. The slope is 12 dB/octave.

1.3 EQUALISER

A three band equaliser with shelving HF and LF sections and a peak/dip MF section with a bell-curve at 3 kHz. The range of all equaliser sections is +/- 12 dB.

1.4 EQ ON

Switch to bring the equaliser section into the audiopath.

1.5 AUX

Extra multi purpose post fader output e.g. for effects equipment, special foldback or PA purposes, in stereo, with levelcontrol.

1.6 PANPOT

Centre detented pot for location of the mono mic/line signal between left and right in the stereo signal. This control is placed before the (stereo) fader, allowing to use the PFL system also for checking the panpot setting.

Below this AUDIO SECTION the FADER AREA starts, using switches and the fader, coupled to the processor control of the module and being flexible in configuration by 8 programming switches.

Starting from the bottom there are:

1.7 ON/OFF BUTTON and -INDICATOR

This is a dual function unit for both control and indication of the channel status. The lamp is fully OFF when the channel is NOT READY, DIM in the READY status and BRIGHT when the channel is ON. Also the lamp blinks on receiving ring voltage on a connected hybrid.
The combination FADER- ON/OFF logic can be configured in 2 ways, determined by the desired mode of operation:

1. "SERIES CONNECTION" of fader and ON/OFF button logic. This means that the channel is open when both the ON/OFF logic is ON and the fader is up. Generating mute signals or studio foldback control is then possible either by the fader or the ON/OFF pushbutton.
2. FADER CONTROL: the fader acts as level setting device and is also used for generating MUTE

commands. The ON/OFF button is only for indication of the channel status.

1.8 FADER

This is a DC analog control for setting the gain of the channel VCA's with a special control law for creating a high resolution working- and a coarse fade out area. The VCA gain can also be influenced by:

1. DC bus for VCA control by limiting
2. Logic control from the processor, like COUGH MUTE.

All "VCA SWITCHING" is done via a rampvoltage for a convenient, fast and silent fade-in or out.
Comparators are used for deriving the logic signals from the fader for position sensing, a mechanical detent can be added for giving the "switch-feel" on opening the fader.

1.9 PFL PUSHBUTTON and INDICATOR

This has always a dual mode of operation:
Depressed short gives a LATCHING action, which means that the PFL status is ON until a second press of the switch or the opening of the channel or activating another channel for PFL. The last mentioned OFF-action can be disabled at wish in the CR-MONITOR module, giving the choice between a MIXED or a SINGLE PFL selection.
Depressed longer makes the PFL function momentary, switching OFF the PFL status on releasing the button.
If the prelistened source is a controlroom mic the louspeakers in the controlroom will be dimmed to prevent howl-round.
The routing of the PFL signal to controlroom loudspeakers, console loudspeakers, headphones and meters is set by the CONTROLROOM MONITOR FUNCTION setting, to be described later.

1.10 COM BUTTON

In case the source is a microphone and the channel is configured as such, this control is for a seperate PFL system, used for COMMUNICATION purposes only. The COM function only works with STUDIO MICROPHONES. When switched ON, closing the channel sends the prefader signal to the COM mix and so to the intercom loudspeaker for a continuous audible contact with selected microphones in the studio.
In case the source is line level being either a music line or the signal coming from a hybrid, this button is meant for communication into the return audiopath (4-wire return line or hybrid return input) available on the connectorpanel of the module.

1.11 INPUT SELECT BUTTON

To be used for selection of the A or B input to the channel. This selection includes not only audio, but also all other logic and VCA control interfaces and remote connections, dependant on the location of the microphones or the line level source connected.
The function is safeguarded by enabling it only with the channel not ON and by depressing the button for longer than 1 second. This to prevent any unwanted change of source in an active channel.
The several types of input sources (dependant also on connector panel) or the location of the microphone connected (CR or STUDIO) can be set by means of PROGRAMMING SWITCHES.

1.12 SEND BUTTON

This can be configured as a switchable postfader output like AUX or as a GROUP ISOLATE ROUTING by a programming switch in the module. The module behaves normal, but all centralised logic- and VCA control is disabled as is the routing to the MAIN outputs. In this setting the SEND signal can be used to make an isolated subgroup for recording of incoming signals while the console is ON AIR or for creating a stereo subgroup, to be mixed into the MAIN signal via a stereo line inputchannel. The SEND routing function can be switched ON or OFF only when the channel is not active for safety reasons.

1.13 FUNCTION BUTTON

This button can have different functions dependant of the type of module and the type of software:
- In the mic mode this is to disable the COUGH for a particular channel.
The LED in this switch also indicates the external COUGH action by blinking.
- In the line level "2-wire" mode this is to enable an auto-answer mode for the hybrid (after 3 times ringing the hybrid is automatically switched to the console so the caller receives the programme via the return output).

All above mentioned pushbuttons have LED indicators and a legending, that can be adapted to the function by simply replacing the inlay in the switchknob. All needed inlays are supplied with the mixer, on request, blanks can be included for e.g. putting the name of a connected source on it.

2. <u>INPUT CONNECTIONS</u>

The connections are located on the back of the mixertray on the connectorpanel.
There are 3 possible arrangements of connectorpanel to be specified at ordering:
- MIC/MIC input sources
- LINE/MIC input sources
- LINE/LINE input sources.
Sources are connected to the channel via XLR connectors. In case of microphone sources each with individual RF filters and phantom powering.
The standard version has a differential inputamplifier, a transformer can be added as an option.

3. <u>INSERTION POINT</u>

The mono channel has a prefader insertion point on a 1/4" jack at a level 3 dB below the nominal outputlevel.

4. <u>RETURN OUTPUT</u>

A second 1/4" jack carries the electronically balanced N-1 mono signal to be used as source for the 2-wire/4-wire return or signal feed to a news booth.

5. <u>CHANNEL REMOTE CONTROL CONNECTOR</u>

Each module has a channel remote control connector, that can be used for extension of the operator interface or for coupling to an automation system. The following signals can be found on the D9 socket:

1. Logic input, programmable to be used as cough input in mic mode or ring-detect input in 2-wire mode.
2. Logic input, programmable to be used as toggle input to be able to bring the channel from the READY status to the ON status by means of an external switch.
3. Logic output coupled to A input selection, suitable for driving LED's or optocouplers, to be used as output signal to turn the hybrid on in 2-wire mode or as a channel on indication in mic or 4-wire mode.
4. Logic output coupled to B input selection, for the same applications as mentioned under 3.

6. PROGRAMMING SWITCHES

Close to the fader 8 programming switches are located on the PCB with the following functions:

SW1 : setting the FADER/ON-OFF behaviour.
SW2 : setting SEND to ISOLATE.
SW3 : setting Input A for mic mode to treat logic for CR mic or ST mic and for line mode to act as 4-wire line source or as 2-wire source.
SW4 : setting Input B for mic and line mode as mentioned under SW3.
SW5 : setting Input A for mic mode to have default the cough function on and for line mode/2-wire mode to have the auto-answer function ON.
SW6 : setting Input B for mic and line/2-wire mode as mentioned under SW5.
SW7 : free for future use.
SW8 : Not used.

D. S342(E) DUAL STEREO LINE INPUT CHANNEL

1. OPERATOR CONTROLS

The function of the individual modules will be described using the layout drawings of the inputchannels. The controls are divided in DIRECT AUDIO CONTROLS in the upper part of the module and the ones in the fader area, coupled to the LOGIC SYSTEM and from there to the audio and logic busses.

The DUAL STEREO LINE INPUT S342(E) has two full function in/outputs, that can be used for the connection of stereo machines or line sources, with full remote control. Also possible is the use of the channel for picking up STEREO MUSIC LINES and using available logic outputs for controlling the 4 wire return signal.

1.1 INPUT GAIN CONTROL

This is a centre detented pot with a range of +/- 10 dB for fine gaincontrol of the channel.
The inputcircuit consists of a differential balanced input. Selection of the A- or B input is via relays, driven by the controller via a frontpanel switch.

1.2 MONO/STEREO MATRIX

Two switches allow for 4 input assignments:

1.	BOTH switches OUT:	the left input to the left channel and the right input to the right channel, the normal STEREO setting.
2.	LEFT switch in :	the left input is connected to both channels (LEFT MONO)
3.	RIGHT switch in :	the right input is connected to both channels (RIGHT MONO)
4.	BOTH switches IN :	a mix of the left and right input is sent to both channels (MIX MONO)

1.3 EQUALISER

Available as an option for the stereochannels.
Consists of shelving HF and LF sections and a peak/dip MF section with a bell- curve at 3 kHz. The range of all equaliser sections is +/- 12 dB.

1.4 EQ ON

Switch to bring the equaliser section into the audiopath.

1.5 AUX

Extra multi purpose post fader output for effects equipment, extra foldback- or PA purposes, in stereo, with levelcontrol.

1.6 BALANCE CONTROL

Centre detented pot for correction of the L/R balance of the channel over a range of +/- 3 dB. This control is located before the fader, allowing the PFL system to be used for checking the balance.

Below this AUDIO SECTION the FADER AREA starts, using switches and the fader, coupled to the processorcontrol of the module and being flexible in configuration by 8 programming switches. Not all possibilities will be mentioned but only the ones now available in the "standard" software. A number of customer specific options can be implemented if the available settings are not adequate for the desired way of operation, also after initial installation.
Starting from the bottom there are:

1.7 ON/OFF BUTTON and -INDICATOR

This is a dual function unit for both control and indication of the channel status. The lamp is fully OFF when the channel is NOT READY, DIM in the READY status and BRIGHT when the channel is ON.Also the lamp blinks on receiving a INHibit fader command from a connected source.
The combination FADER- ON/OFF logic can be configured in 2 ways, determined by the desired mode of operation:

1. "SERIES CONNECTION" of fader and ON/OFF button logic. This means that the channel is open when both the ON/OFF logic is ON and the fader is up. Generating machine control signals or studio foldback control for microphone mixers is then possible either by the fader or the ON/OFF pushbutton.

2. FADER CONTROL: the fader acts as level setting device and is also used for generating faderstart and -stop commands. The ON/OFF button is only for indication of the channel status. This is the normal "European way" of using a broadcast desk.

1.8 FADER

This is a DC analog control for setting the gain of the channel VCA's with a special control law for creating a high resolution working- and a coarse fade out area. The VCA gain can also be influenced by:

1. DC bus for VCA control by overall limiting.
2. Control from a connected machine (on A-input only): if a machine is in the RECORD status, the channel will be closed automatic to prevent after tape signals to enter the mix if the channel was open by accident. After tape listening is than possible using the PFL function.

All "VCA SWITCHING" is done via a rampvoltage for a convenient, fast and silent fade-in or out.
Comparators are used for deriving the logic signals from the fader for position sensing, a mechanical detent can be added as an option for giving the "switch-feel" on opening the fader.

1.9 PFL PUSHBUTTON and INDICATOR

This has always a dual mode of operation:
Depressed short gives a LATCHING action, which means that the PFL status is ON until a second press of the switch or the opening of the channel or activating another channel for PFL. The last mentioned OFF-action can be disabled at wish in the CR-MONITOR module, giving the choice between a MIXED or a SINGLE PFL selection.
Depressed longer makes the PFL function momentary, switching OFF the PFL status on releasing the button.
As an extra feature START- and STOP signals can be generated by the PFL action, for easy cueing of source machines from the desk. This can be set by a programming switch per input and is only available on the module PFL button and only possible with the channel OFF or READY. Normal PFL without start coupling is possible in all states of the fader.

Both PFL switch and -LED are also available on the machine in/out connector for remote control of the function, individual for the A- selection.
An extra function is given to the EXTERNAL PFL inputs: when the channel is OFF and the input is selected to the B input, depressing the EXTERNAL PFL button longer changes over the channel to the A-input.

The routing of the PFL signal to controlroom loudspeakers, console loudspeakers, headphones and meters is set by the CONTROLROOM MONITOR FUNCTION setting, to be described later.

1.10 ST BUTTON

This is a multi purpose pushbutton, which output is available on the machine connection, individual for the A- and B input. It can be used for extra remote control functions from the desk like blind start, stop, zero-locate, FFW or locate for CD players.

1.11 INPUT SELECT BUTTON

To be used for selection of the A or B input to the channel. This selection includes not only audio, but also all other machine interfaces and remote controls.
The function is safeguarded by enabling it only with the channel not ON and by depressing the button for longer than 1 second. This to prevent any unwanted change of source in an active channel.
Remote control of the input selection is also possible via the EXTERNAL PFL inputs, as described under PFL.

1.12 SEND BUTTON

This can be configured as a switchable postfader output like AUX or as a GROUP ISOLATE ROUTING by a programming switch in the module. The module behaves normal, but all centralised logic- and VCA control is disabled as is the routing to the MAIN outputs. In this setting the SEND signal can be used to make an isolated subgroup for recording of incoming signals while the console is ON AIR or for creating a stereo subgroup, to be mixed into the MAIN signal via a stereo line inputchannel. The SEND routing function can be switched ON or OFF only when the channel is not active for safety reasons.

1.13 FUNCTION BUTTON

This button can have different functions dependant of the type of module and the type of software. In the stereo modules e.g. it determines if the timer should be coupled to this module and with this input selection.

All above mentioned pushbuttons have LED indicators and a legending, that can be adapted to the function by simply replacing the inlay in the switchknob. All needed inlays are supplied with the mixer, on request, blanks can be included for e.g. putting the name of a connected source on it.

2. <u>UNIVERSAL MACHINE INTERFACE</u>

The connection of source equipment to the stereochannels is by means of multipole D25 sockets and 12 pair cable, individual for the A and the B input. The following in- and outputs are available when the channel is set for use with STEREO LINE sources:

1. RECORD RIGHT: a balanced signal for recording the MAINR mixer output.
2. RECORD LEFT : the same, but the MAINL signal. Record signals are parallel connected to both the A and B connector.
3. GROUND : used as reference.
4. INPUT LEFT : left channel input, balanced.
5. INPUT RIGHT : right channel input, balanced.
6. PFL SWITCH : (A-input only) an input from a PFL button to be located at the machine for remote control of the channel PFL and input select function. It can be used this way for remote after tape checking of recording machines or for cueing playback machines.
7. PFL LED : (A-input only) indication of the activation of the PFL function.
8. START : a continuous or momentary signal on opening the channel or any programmed start function.
9. STOP : a momentary signal on closing the channel or any programmed stop function.
10. KEY : free to use remote control output, derived from the ST button.
11. INHibit fdr : (A-input only) logic input from a connected machine for controlling the channel logic, for closing the channel when a connected recording machine comes in the RECORD status.

All control outputs are isolated by optocouplers to prevent any unwanted ground links in the studio system. Also the FDRINH input is made via an optocoupler for the same reason.

3. PROGRAMMING SWITCHES

Close to the fader 8 programming switches are located on the PCB with the following functions:

SW1 : setting the FADER/ON-OFF behaviour as described under C.1.7.
SW2 : setting the SEND output to RECORD ISOLATE.
SW3 : setting the START A output to pulse or continuous.
enabling or disabling the STOP A pulse output.
SW4 : setting the START B output to pulse or continuous.
enabling or disabling the STOP B pulse output.
SW5 : enabling the timer with input A selected and fader opened.
SW6 : enabling the timer with input B selected and fader opnened.
The timer enabling/disabling can be altered by the FUNCTION pushbutton on the front.
SW7 : PFL function coupled to START control of input A enabling.
SW8 : Not used.

E. STUDIO MONITOR MODULE S347

This module is meant for control of loudspeakers, presenter- and guest headphones in the studio. The functions are divided over three physical units:

- the input source selection (for "internal" and "external" sources) is located on the top part of the mainboard,
- the switches are mounted on a seperate PC board parallel to the frontpanel,
- the audio- and coupled control-circuits are combined with the controls on the lower part of the S347 module.

1. <u>OPERATOR CONTROLS</u>

1.1 STUDIO MONITOR SOURCE SELECTION

6 pushbuttons with LED's are used for selection of the monitor source.
The following selections are available:

1. MAIN output.
2. SEND B output, in most ON-AIR applications this is used as selective (N-mic's) studio foldback.
3. SEND (C) output.
4. AUX output.
5. EXTernal stereo source 1.
6. EXTernal stereo source 2.

The DEFAULT inputselection when switching the console on is MAIN.

1.2 STUDIO MONITOR FUNCTION SELECTION

2 internal jumpers are available for selection of the behaviour of the presenters headphone signal on activating the TALKBACK function. In Normal mode the original signal is dimmed and the commands appear on both sides of the headphone; in Split the original signal comes on the RH side of the headphone and the command appears on the LH side.
An extra jumper next to the EPROM marked SELFB can be set when SEND B is used as selective studio FB (N-mic's in that studio). The HPH's change over to MAIN when a studio mic becomes live although SEND B is selected. This rather safe signal will continue on the studio speakers.

1.3 STUDIO LOUDSPEAKER FUNCTIONS

The studio loudspeakers will follow the monitor selector and are overridden by talkback when pressing the command button marked STUDIO on this module (or external by the TBR button). The level is determined by the pot marked LSP. The speakers are muted when a studio mic becomes live unless SEND B, set as selective studio foldback, is selected (and the jumper is placed).

1.4 GUEST HEADPHONES FUNCTIONS

The GUEST outputs are always connected to the SELection of sources, interrupted by talkback when the command button marked STUDIO is pressed, or MAIN (see 1.2) when a mic is live in the studio.
The (stereo) output is suited for most common types of headphones, preferably of medium to high impedance for the best results. The level is set by the pot marked GUEST.

1.5 PRESENTER HEADPHONES FUNCTIONS

The presenters headphones normally follow the same signal as the guest headphones but can be individually adressed by means of the command button marked PRES. The headphones give the talkback on either one or both sides (see 1.2).
The (stereo) output is suited for most common types of headphones, preferably of medium to high impedance for the best results. The level is set by the pot marked PRESENTER.

2. <u>CONNECTIONS</u>

INPUTS for the external sources are the same as the ones on the ControlRoom Monitor module and routed to the toprack where they can be either wired to XLR's or to other selector banks as extension.All other signals, such as LOUDSPEAKER outputs, HEADPHONE outputs (pres and guest), signalisation output and the cough input are on 25 pin D type connector to be wired to the studio with one multicore cable.

F. CONTROLROOM MONITOR MODULE S348

This module is meant for control of loudspeakers, operator- and guest headphones in the control room. The functions are divided over three physical units:

- the input source selection (for "internal" and "external" sources) is located on the top part of the mainboard,
- the switches are mounted on a seperate PC board parallel to the frontpanel,
- the audio- and coupled control-circuits are combined with the controls on the lower part of the S348 module.

1. OPERATOR CONTROLS

1.1 CONTROL ROOM MONITOR SOURCE SELECTION

6 pushbuttons with LED's are used for selection of the monitor source.
The following selections are available:

1. MAIN output.
2. SEND B output, in most ON-AIR applications this is used as selective (N-mic's) studio foldback.
3. SEND (C) output.
4. AUX output.
5. EXTernal stereo source 1.
6. EXTernal stereo source 2.

The DEFAULT inputselection when switching the console on is MAIN.

1.2 CONTROLROOM MONITOR FUNCTION SELECTION

1 internal jumper is available for selection of the behaviour of the presenters headphone signal on activation of the TALKBACK function. In Mixed mode the original signal is dimmed and the commands appear on both sides of the headphone; in Split the original signal comes on the RH side of the headphone and the command appears on the LH side.

4 extra jumpers next to the EPROM marked F1-F4 can be set to get the several possible monitor modes (the marked position is the OFF position and gives the first mentioned mode):

F1: PFL single or mixed. (Selecting PFL resets another or not)
F2: PGM mode OFF or ON. (Speakers always give source selection; PFL only on little speaker).
F3: Main monitor Split or Normal. (When selecting PFL, it comes in mono on one speaker only (S) or in stereo (N)).
F4: COMDC enabled or disabled at CR mute. (When a CR mic is live the operator can not be disturbed by commands or intercom).

1.3 CONTROL ROOM LOUDSPEAKER FUNCTIONS

The Controlroom loudspeakers will follow the monitor selector and are overridden by PFL (in stereo or mono; see 1.2) when any of the PFL buttons on the input-channels is pressed.
The output is muted when a Controlroom mic becomes live and is dimmed when any of the command buttons on the console is pressed. The level is determined by the pot marked LSP.

1.4 OPERATOR HEADPHONES FUNCTIONS

The operator headphones normally follow the same signal as the speakers being monitor selector or PFL, but can be interrupted by any COM signal coming in. The headphones give this signal on either one or both sides (see 1.2). The level is set by the pot marked OPERATOR.
The (stereo) output is suited for most common types of headphones, preferably of medium to high impedance for the best results.

1.5 GUEST HEADPHONES FUNCTIONS

The GUEST headphone outputs can be selected to give either the source selector or main output to to be set by means of jumpers marked GUEST.
The (stereo) output is suited for most common types of headphones, preferably of medium to high impedance for the best results. The level is set by the pot marked GUEST.

2. CONNECTIONS

INPUTS for the external sources are the same as the ones on the Studio Monitor module and routed to the toprack where they can be either wired to XLR's or to other selector banks as extension.
LOUDSPEAKER outputs are on two male XLR's,
HEADPHONE outputs are on 1/4" stereo jacks.
Signalisation output for CR signalling can be found routed to the toprack.

G. MASTER MODULE S346M

This module houses all mix- and line amplifiers for all balanced and unbalanced audio signals.
Also connections are made via interfaces to the toprack for both audio- and logic signals that can be needed there. These signals are all located on a CENTRAL DISTRIBUTION PANEL in the toprack, where they are all available on multiple pins for an easy wiring to the appropriate EA 700 function modules.
Also in this module the talkback mic and preamp is located.
Output connections (MAIN, MONO, RECORD, OUTB, OUTC and AUX) are to be found on a 25 pin D type connector.

H. SCRIPT SPACE S346S

This area, which takes up the place for 6 normal modules, offers space for a A4 size script and also gives the opertunity to convert the 25 pin D type connector with outputsignals to 12 individual 3 pin male XLR's as well as 4 female XLR's for the external monitor inputs all on a single connector panel.

I. TOPRACK: EA700 SERIES MODULES

A great deal of the flexibility of the S340 design is based on the use of Eurocard modules from our EA700 series. These units add a lot of external facilities to the mixer.
In the mixer all necessary audiosignals and control inputs and -outputs are generated for driving the EA700 modules.
The range of available units is growing with the number of delivered consoles; each customer has his own way of working isn't it?

Available as more or less standard units are now:

EA700/1 Blank panel 40 mm.
EA700/2 Blank panel 80 mm.
EA704 Console loudspeaker module with built-in amplifier
EA706/1 Intercom unit for maximum 7 destinations
EA706/2 Intercom unit for maximum 13 destinations
These units work via a relay matrix/switcher.
EA709 Dual 19 segment LED PPM.
EA710 Oscillator.
EA712/1 Real time clock for receiving the DCF77 signal
with built-in timesignal generator.
EA717/1 Studiotimer/stopwatch with preset and countdown.
Available in more versions.
EA718 Control line interface, receiving.
EA719 Control line interface, sending.
EA721 Compressor/limiter for overall limiting on all channel-VCA's.
EA732 Output (master) module for AUX/SEND signals.
EA756 6 way source selector, passive.
EA757 12 way source selector, passive.
EA760 Metermodule, available in more versions (type meters).

Other additions to the S340 system are analogue or digital telephone hybrids, electronic crosspoint switchers, red light power switches and other units available in our EA800 or EA900 series.

As an example, some of the frontpanel layouts are supplied.