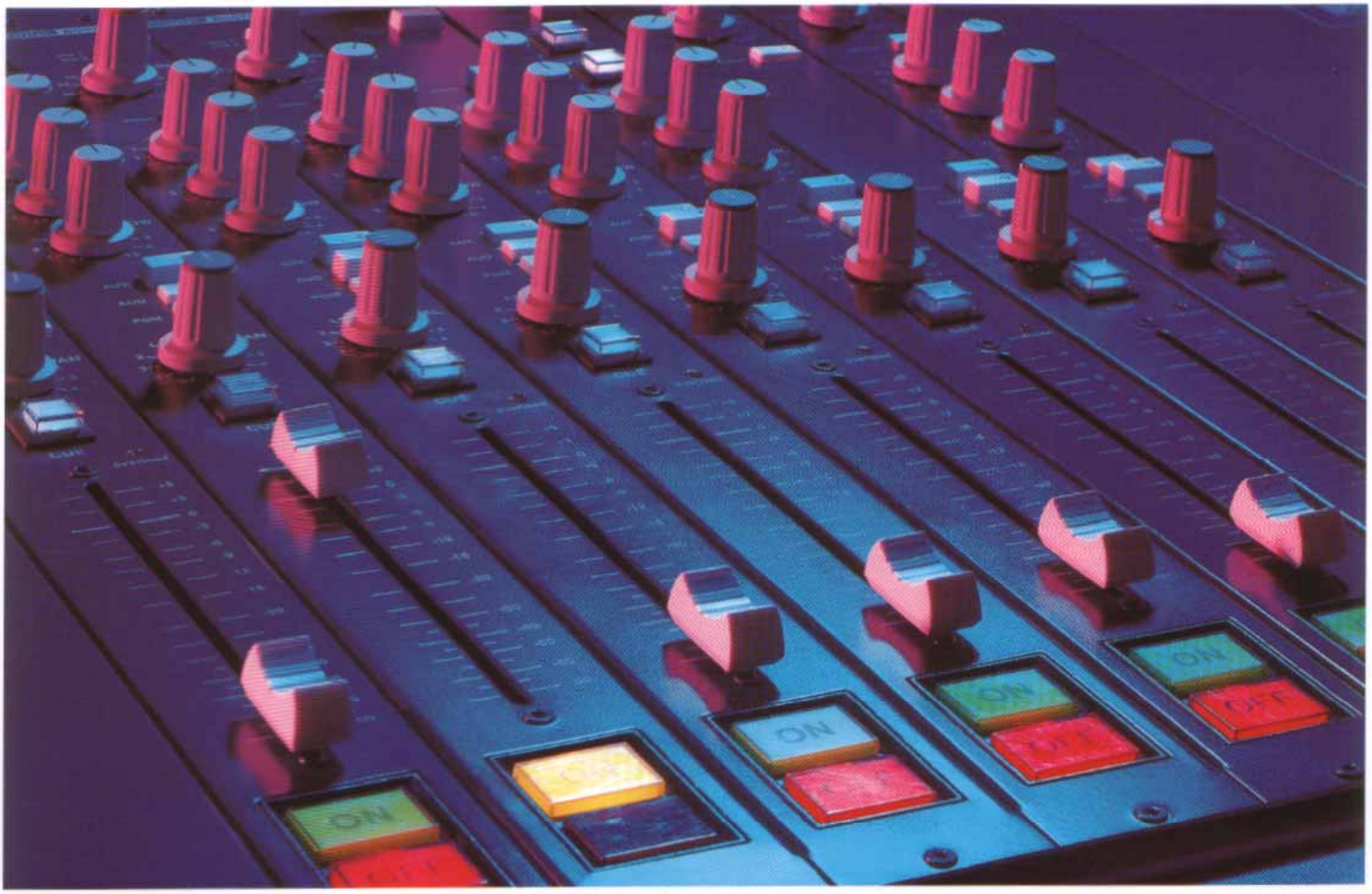
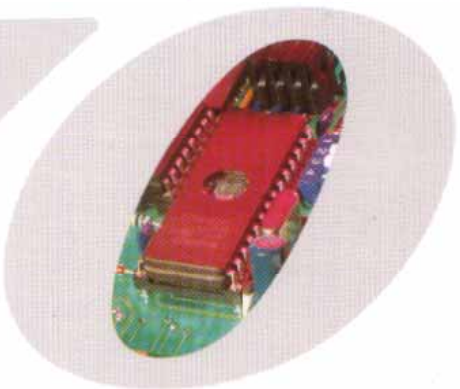


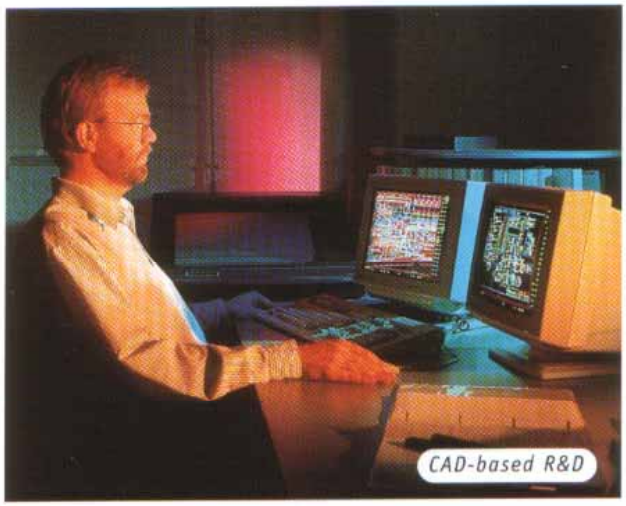


DATEQ

BCS BROADCAST CONSOLE

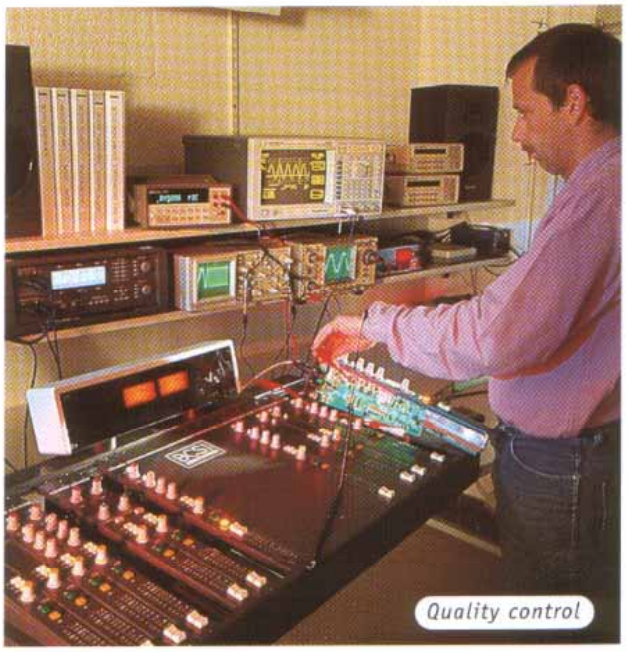
Dateq BCS 70

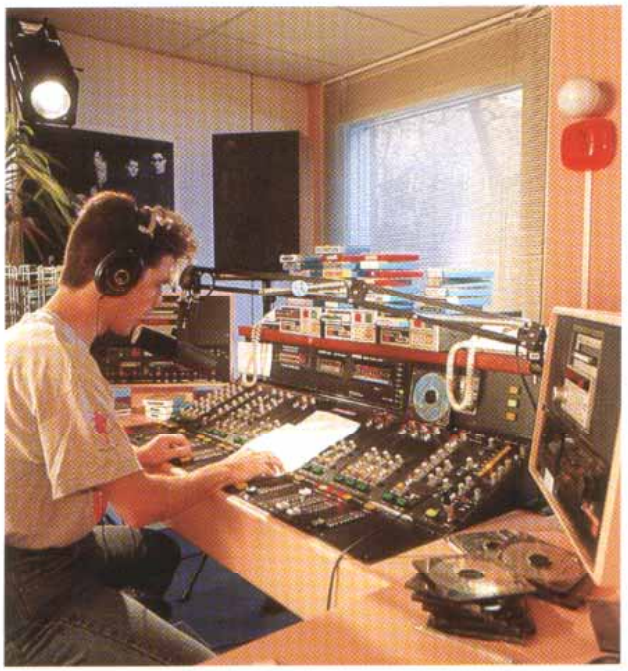
The technology of broadcasting is changing rapidly. The number of auxiliary equipment is growing exponentially, creating a multitude of connection possibilities. Above all, demands on the communication between studio and outside world are getting higher and higher. All these problems


CAD-based R&D

can only be solved with a totally flexible broadcast console.

Dateq originally started in 1970 making audio equipment for DJ's and other professional users. The first mixing console series was developed in close co-operation with those end-users and became a huge success. In 1984 the Dutch broadcast market was opened to other (non-government-run) stations: this has revolutionised the market. That same year Dateq introduced the first broadcast console, based on the knowledge and state-of-the-art technology from the original mixing console designs. Since 1984 a lot has changed, but one thing remained: our philosophy to develop mixing consoles in close co-operation with the end-user while, at the same time, giving maximum priority to overall quality and serviceability.

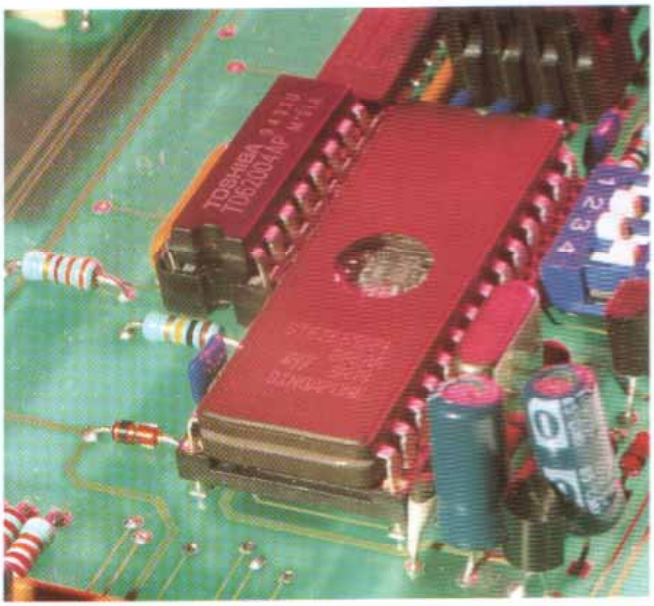

Quality control



"A new way of broadcasting" is an apt description of the BCS 70 broadcast console. The powerful microprocessor control-system is just one of the features making this console so flexible, easy to use and above all: intelligent. The unique studio intercom function is specifically designed to cope with the ever-increasing communication demands of the modern radio-studio. All this combined with the unique user-friendliness you have come to expect from any Dateq console.

A new way of broadcasting

The BCS 70's foundation is formed by the modular frame which may hold up to eighteen modules. Any mix of input modules is possible, so the user can custom-configure any console. The standard frame is empty, except for a script-space, a master module and an output section. The BCS 70 frame can also be supplied in a split-console configuration. The console may be used with or without the optional meter bridge.



The power supply is mounted in a separate (external) 19"-housing.

Microprocessor control

The most unique feature of the BCS 70 is the microprocessor control system which has command over all audio circuits, routing and switching. This is the reason why the BCS 70 is so extremely easy to use and what gives it the optimised ergonomic design. It also makes remote control of auxiliary equipment much more powerful. In addition to the usual latching or continuous type start/stop contacts, a number of equipment-specific connections is also possible. For instance, equipment CUE, READY or PLAY control signals can activate the indicator lights in the large START/STOP buttons - which are conveniently placed below the faders. The microprocessor memory holds settings for a great number of replay equipment in actual use today. And, of course, you can have Dateq add all your equipment-specific settings to the memory...

Connections

Each input channel has two separate balanced inputs, each with their own logic and gain control. A 25-pin Sub-D connector handles all signals to and from the module, including those from external equipment (tally, remote control, etc.). The channel insert is balanced also. The BCS 70 has a large number of balanced XLR outputs. The (left, right and mono) PGM, AUD and AUX outputs are positioned below the script space on the back panel. The master module features the Control Room outputs (XLR-balanced), the two ON AIR group outputs and a connector for the Program Director Station (intercom). Additionally there are balanced inputs for the AIR signal and two 'spare' signals.

Input Modules

Each input module uses a high-quality conductive plastic fader to control the VCA (Penny&Giles fader is optional). The VCA is of the newest generation with drastically reduced distortion specs. The signal-to-noise ratio is very high, making the console extremely 'silent'. Each input module has three routing switches, thereby making it possible to do pre-production work during the actual transmission.

BCS 70
broadcast console

The BCS 70 broadcast console

Dateq BCS 70

Holds up to eighteen modules

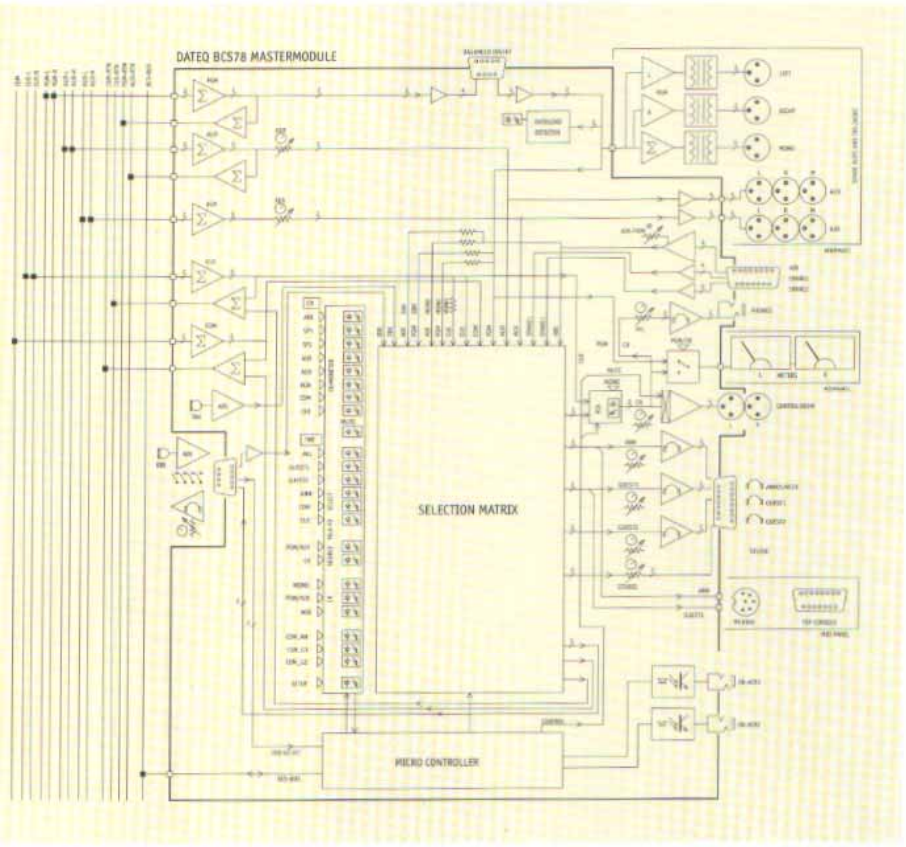
- The BCS 70 frame holds up to eighteen modules
- Each module has two balanced inputs, so switching between two input signals is easy
- Each module has balanced inserts
- Each module has its own microprocessor controlling audio routing, channel settings and external equipment
- There are modules with and without equalizer and/or gain control
- Three band equalizer for precision tailoring of the sound
- Routing switches. The AUX routing can be selected pre- or post-fader
- Conductive plastic faders for VCA-control
- Ergonomically positioned, large illuminated START/STOP buttons
- The button lights are controlled by the module (internal tally) or by the connected equipment (external tally)

- Extensive communication facilities between guests, engineer and presenter plus talkback to the telephone channels
- SET-UP button to program the most common control functions
- BCS 81 clock/timer module
- BCS 68-2 optional 30 segment LED stereo VU-meter with peak-hold function
- Meter bridge with either analogue VU-meters, 50 segment LED Peak Program Meters or a 200 segment orange neon gasdischarge display PPM

Two separate balanced inputs

Dateq BCS 70

The Master section has a completely new routing concept to the outputs, using a microprocessor and an audio signal matrix.



SET UP Selects SET UP mode for programming functions

STUDIO Gain control for studio output

METER PGM/C.R. Selects the output meters between PGM or Control Room output signal

AUX Gain control for AUX output

AUD Gain control for AUD output

AIR ... CUE Multisource select buttons for the C.R. output signal

MUTE LED to indicate that C.R. monitors are switched off

C.R. Gain control for C.R. output

OVERLOAD LED to indicate overload of Master section

MONO Switches the C.R. output signal to mono

PGM/AIR Selects whether C.R. output signal switches to PGM or AIR signal, as soon as the DJ MIC channel is 'opened'

MIX Selects whether PGM or AIR signal is mixed (at lower level) with the CUE signal

PHONES Gain control for headphone output

Master Module

Unique to the Master section are the COM buttons for Guest1, Guest2 and the Announcer, next to their respective output GAIN controls. Depressing a button will send that output signal to the COM mix-buss. This is very convenient, for instance, when during the program a guest would like to continue a telephone conversation with a listener, off-air. Or when the presenter would like to talk to reporters on location, without using the CUE-function and mix-buss. After switching the reporter's channel ON AIR, he or she will hear the PGM signal return. Switching off-air again, he or she is automatically returned to the COM buss signal feed.

The eight Multisource Select buttons select which signal will be heard from the control room monitors (AIR ... CUE). These same buttons are also used during SET UP mode to program the settings of the console (using the blue text markings). You can, for instance, select each module between fader- or button-start or whether the CUE button (pre-fade-listen) also changes the monitor mode (from speakers to phones or vice versa). You may select microphone channels to a different ON AIR group, so the correct ON indication will light when a microphone is 'opened'. Or you may select different mute-settings. The additional button is used to select whether the Timer will be controlled from specific modules.

Dateq BCS 70

COM (GUEST 1) Selects the GST1 signal to COM buss

GST Gain control for GST1 output

COM (GUEST 2) Selects the GST2 signal to COM buss

GST 2 Gain control for GST2 output

ANN Gain control for ANN(ouncer) output

COM (ANNOUNCER) Selects the ANN signal to COM buss

TALK BACK/SRC Selects the signal (talkback or source) to the ANN, GST1 and GST2 output

MIC Electret talkback microphone

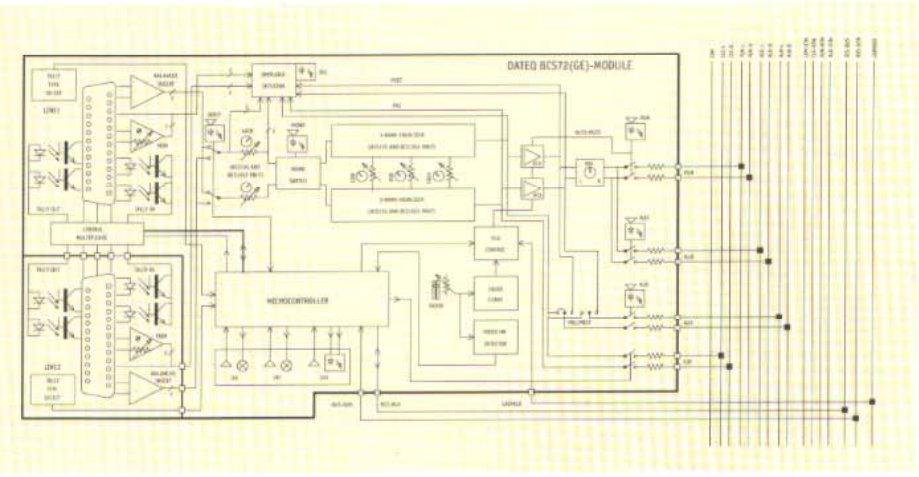
TALK ALL ... TALK CUE Communication select buttons route talkback signal to GST1, GST1, ANN, COM and/or CUE (more than one is possible)

PHONES Stereo headphone output jack

Master Module

Dateq BCS 70

The stereo input module has two identical line level inputs, so you may connect two different signal sources to one module. In addition to these balanced line inputs, each module has a balanced insert point on each input plus switching contacts for remote control of external (replay-)equipment.



The BCS 72 Stereo Module is available in the following versions

BCS 72	line/line input
BCS 72E	line/line input with three-band equalizer
BCS 72G	line/line input with gain control
BCS 72GE	line/line input with gain control and three-band equalizer

LINE/LINE Selects the correct input (Line1 or Line2)

GAIN Gain control

MONO Switches stereo input signal to mono

HIGH High frequency equalizer

MID Mid frequency equalizer

LOW Low frequency equalizer

AUX Routes input signal to AUX buss.

AUD Routes input signal to AUD buss

PGM Routes input signal to PGM buss

BAL Balance control for left and right outputs

CUE Pre-fader-listen via Control Room monitors or headphones

OVERLOAD LED indication for signal overload

FADER Conductive plastic fader for VCA-control

ON/OFF Switches channel on or off and starts/stops external equipment

Stereo Module

Dateq BCS 70

MIC/LINE Input select switch for MIC input ('out') or LINE input ('in')

GAIN Gain control for MIC and LINE input

LOW CUT Selects Low Cut filter between 20 Hz ('out') or 80 Hz ('in')

HIGH High frequency equalizer

MID Mid frequency equalizer

LOW Low frequency equalizer

AUX Routes input signal to AUX buss

AUD Routes input signal to AUD buss

PGM Routes input signal to PGM buss

PAN Pan control: positions input signal in stereo image

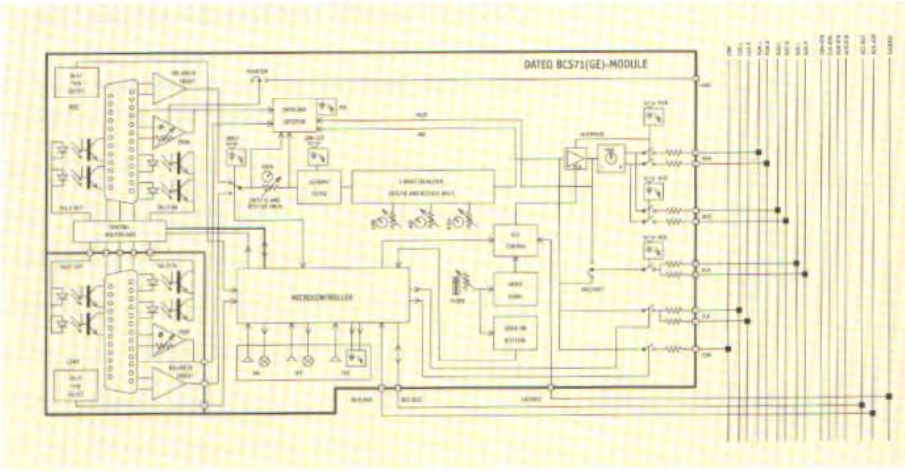
CUE Pre-fader-listen via Control Room monitors or headphones

OVERLOAD LED indication for signal overload

FADER Conductive plastic fader for VCA-control

ON/OFF Switches channel on or off and starts/stops external equipment

This module can handle one microphone signal plus one line level signal. The BCS 70 may be used with four different Mono Module versions.



The BCS 71 Mono Module is available in the following versions

BCS 71	mic/line input
BCS 71E	mic/line input with three-band equalizer
BCS 71G	mic/line input with gain control
BCS 71GE	mic/line input with gain control and three-band equalizer

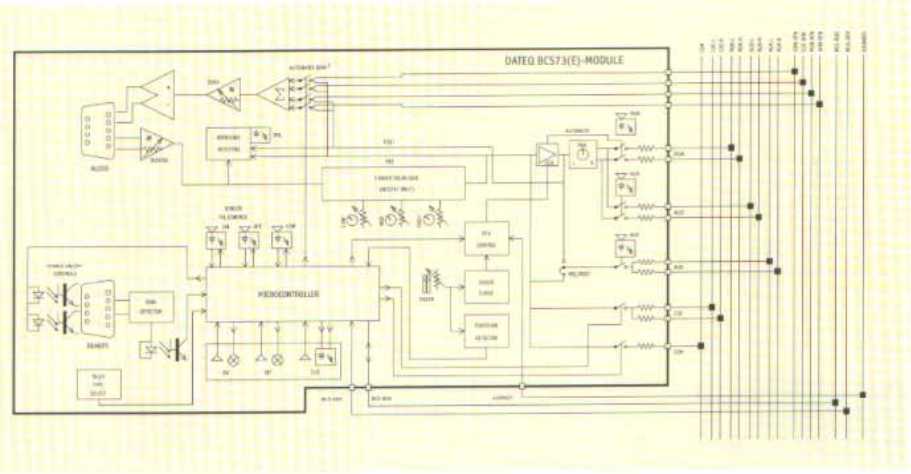
With the AUX, AUD and PGM buttons you may route the audio signal to three separate outputs. During transmission you can, for instance, route a telephone conversation to another buss output for recording purposes.

The phantom power can be set through an internal jumper, the pre-/post-fader selection for the AUD/AUX buss is also set via jumpers.

Mono Module

Dateq BCS 70

Dateq's QDM2 (Quad Dynamical Mix Minus) is a highly advanced telephone routing system. With this system each telephone module can generate its individual clean-feed return with the source signal removed. This obviates the need for separate clean-feed mix busses, so the number of

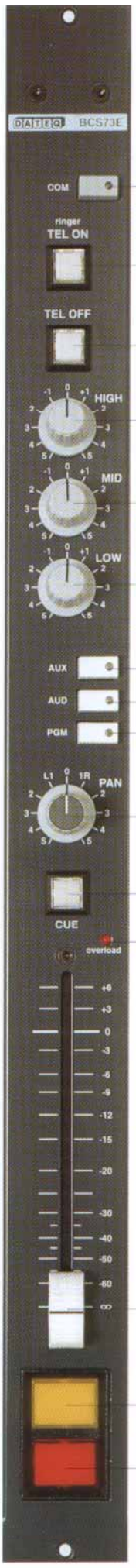


usable telephone hybrids is theoretically limitless! Each telephone channel can generate a Program Audition, Cue (pre-fade-listen) or Communication buss signal as return feed.

Telephone Module

The BCS 73 Telephone Module is available in the following versions

BCS 73	with remote control for external telephone hybrid
BCS 73E	with remote control for external telephone hybrid and three-band equalizer



COM Routes input signal to communication buss

TEL ON/OFF Controls external hybrid (ring detection by light in switch)

HIGH High frequency equalizer

MID Mid frequency equalizer

LOW Low frequency equalizer

AUX Routes input signal to AUX buss

AUD Routes input signal to AUD buss

PGM Routes input signal to PGM buss

PAN Pan control: positions input signal in stereo image

CUE Pre-fader-listen via Control Room monitors or headphones

OVERLOAD LED indication for signal overload

FADER Conductive plastic fader for VCA-control

ON/OFF Switches channel on or off

The BCS 70 broadcast console can be supplied with one of three types of meter bridge, a clock/timer module, an external LED-meter or a specially designed Program Director Intercom Station .

Meter bridges

The user can select one of three meter bridges: analogue VU-meters, 50 segment LED bar graph VU meters with peak hold or a 200 segment orange neon gasdischarge display PPM (DIN spec.).

BCS 68-2 30 segment LED meters

May be used if an extra pair of meters is needed. For instance, in a production room where the actual equipment is placed out-of-sight.

BCS 81 DCF clock/timer module

The BCS 81 Clock module uses the atomic clock-time as transmitted by the DCF-77 radiostation in Mainflingen, Germany. In addition to the unsurpassed accuracy, the clock automatically switches to Daylight Savings Time (and back) and utilises any 'leap seconds' transmitted. The timer may be used in count-up, count-down or count-up/-down mode and can be set with the supplied remote control.

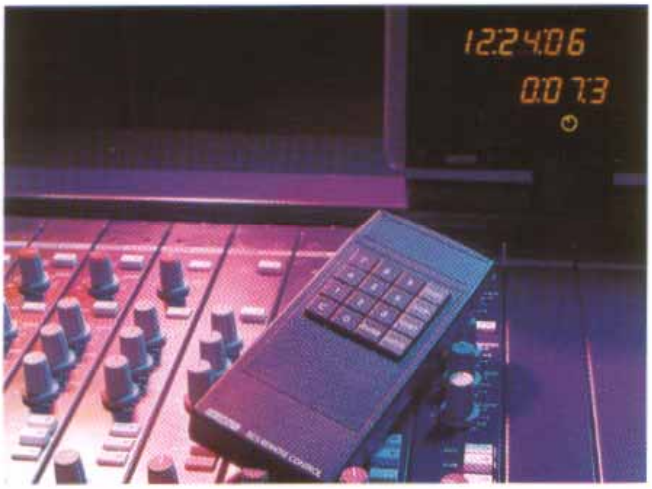
Program Director Station

Communication in and around the studio is very important. The program director is responsible for the final result and may, during transmission, regularly need to communicate with engineer, presenter and/or guests. Additionally he or she has to pre-screen all incoming telephone conversations and may want to pre-check signal sources, the AIR signal, etc. The BCS 70 Program Director Station is specifically developed to solve these problems, to free the engineer and, at the same time, to give the program director complete operational flexibility.

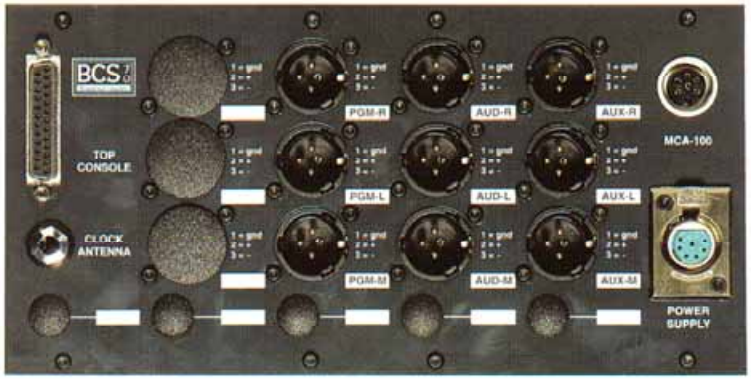
The Program Director Station comes standard with:

- built-in microphone and loudspeaker for communication and signal monitoring
- two stereo headphone output jacks
- one microphone input jack
- 24 buttons for source selection, talkback and control of two BCS 73(E) modules.

An additional eight-button-block can be added for controlling up to two extra BCS 73(E) modules.



Meter bridge, clock/timer module & intercom unit



INPUTS (with VCA-control)

MIC		electronically balanced, 300 Ω (nominal)
	level	-54 ...+14 dB pre-trimmer
		-12 ...+12 dB gain control (gain version only)
	noise	<-127 dB
LINE		electronically balanced, 44 kΩ (nominal)
	level	-14 ...+26 dB pre-trimmer
		-12 ...+12 dB gain control (gain version only)
	noise	<-80 dB
INSERT		electronically balanced, 44 kΩ (nominal)
	level	+6 dB
AIR		electronically balanced, 44 kΩ (nominal)
	level	-20 ... +6 dB pre-trimmer
SPARE1, SPARE2		electronically balanced, 44 kΩ (nominal)
	level	0 dB

OUTPUTS

PGM left/right/mono		transformer balanced
	level/impedance	+6 dBm @ 600 Ω
INSERT		unbalanced
	level/impedance	+6 dBm @ 600 Ω
AUD left/right/mono		electronically balanced
	level/impedance	+6 dBm @ 600 Ω
AUX left/right/mono		electronically balanced
	level/impedance	+6 dBm @ 600 Ω
CRM left/right		electronically balanced
	level/impedance	+6 dBm @ 600 Ω
ANN, GUEST1, GUEST2		stereo headphone, variable
	level/impedance	2x 0.5 W @ 4 Ω
STUDIO		unbalanced, variable
	level/impedance	+6 dBm @ 600 Ω

LOW CUT FILTERS

MIC		12 dB/octave
	-3 dB point	switchable between 20 or 80 Hz
LINE		12 dB/octave
	-3 dB point	10 Hz, fixed

EQUALIZER ('E' and 'GE' version only)

MIC/LINE mono		
	HIGH	+/-12 dB @ 12 kHz, shelving
	MID	+/-12 dB @ 1.3 kHz, bell
	LOW	+/-12 dB @ 60 Hz, shelving
LINE stereo		
	HIGH	+/- 12 dB @ 12 kHz, shelving
	MID	+/- 12 dB @ 1.3 kHz, bell
	LOW	+/- 12 dB @ 60 Hz, shelving
TEL		
	HIGH	+/- 12 dB @ 3.5 kHz, shelving
	MID	+/- 12 dB @ 1.3 kHz, bell
	LOW	+/- 12 dB @ 600 Hz, shelving

GENERAL

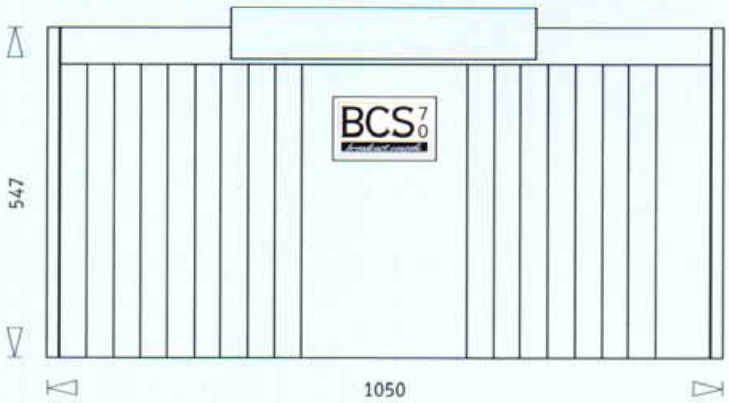
Frequency response		
	Mic to master	20 Hz - 30 kHz, +0/-0.5 dB
	Rest to master	20 Hz - 30 kHz, +0/-0.5 dB
THD + IM		0.02% nominal
Cross-talk L <> R		<-73 dB @ 1 kHz
Cross-talk 1 <> 2		<-80 dB @ 1 kHz
Noise		<-100 dB (inputs OFF)
		<-80 dB (inputs ON)
Overload indication		6 dB below clip level
Clip level		+21 dB

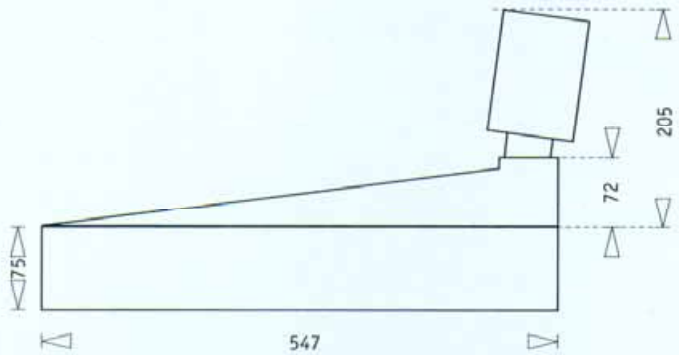
CONTROL I/O (tally)

Tally inputs	opto-coupler, 5 ...15 VDC
Mute button	22 kΩ, normally off
Remote	opto-coupler, max. 25 VDC, 20 mA
ON AIR lamp output	12 VDC, 1 VA
ON AIR outputs	opto-coupler, max. 25 VDC, 20 mA
	solid-state driver, 12 VDC, 10 mA
Remote buss	BCS 70 protocol
Power supply	external supply unit

* DATEQ Audio Technologies reserves the right to amend specifications without notice in line with technological developments. Reference level 0 dB = 0.775 V

MEASUREMENTS (in mm)





Dateq Audio Technologies B.V.
De Paal 37, NL-1351 JG Almere
Tel.: (+31) 36 54 72 222
Fax: (+31) 36 53 17 776
E-mail: info@dateq.nl

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