analog Broadcast Mixing Consoles BC4

BC-IM14

Broadcast Mono Input Module

The BC-IM14 is the most important input module of the BC4 broadcast console system. It offers:

- **Transformer balanced microphone preamplifier**
- **Separate line input amplifier, electronically balanced or transformer balanced**
- **10 auxiliaries, six monos and 2 stereos with level and pan**
- **24 dB/oct. sweeped high-pass filter**
- **Routing section for the selection of 8 sub groups and Two stereo program masters**
- **Balanced, switched, insert pre or post equalizer**
- **4 band EQ with two fully parametric mid bands and sweeped high and low bands**
- **100 mm conductive plastic VCA fader**
- **6 VCA groups**
- **Sophisticated remote control system for external equipment and speaker microphones**
- **Extensive PFL/AFL system**
- **10 LED peak level meter.**

The BC4 Broadcast Console System by adt-audio in Germany covers the range of medium to large format on-air and production consoles for any kind of applications. The rich feature set includes all special functions for broadcasting.

The BC4 console system combines high reliability, long lifespan, and professional technical qualities in combination with excellent sound performance,

In addition to a couple of standard input, group, and master modules, BC4 is a versatile base for custom build broadcast consoles at affordable prices. The system limits offer the choice to makes custom modules that use up to 36 bus rails, which can be used as main masters, group masters and sends in many different ways.

The BC-IM14 has two totally separated inputs. The **MICROPHONE INPUT** is balanced and floating. The high quality input transformer, brand Haufe, Germany, offers excellent CMRR of more than 70 dB @ 15 kHz. The pre amplifier uses adt-audio's double balanced technique that results in excellent noise performance not only with high gain settings but also in the gain range of 40 dB that is mostly used in real world applications. The gain control pot covers the range from 25 dB to 70 dB. In addition there is an input pad, pre the transformer that makes possible to use the mic input for line level sources as well. The standard attenuation of this pad is 25 dB; however, any other required value can be implemented without additional cost.

The **LINE input** has its own input amplifier and gain control stage that operates independent of the microphone preamplifier to maintain best possible cross talk values, also with high gain settings of the mic preamp. The gain control range of the line input is +/- 20 dB. The center detent of the gain pot is internally calibrated. The line input can be implemented electronically balanced or balanced and floating, using a high quality line level input transformer, brand Haufe, Germany. Both options offer high input impedance of more than 10 kOhms and more than more than 60 dB CMMR @ 15 kHz. The design makes it possible to upgrade the electronically balanced version with a transformer at any time.

The default input is the microphone input. The LINE switch selects the line input. In addition, it is possible to rout the console oscillator to the input by the OSC switch. The phase reversal switch operates on the selected input.

The BC-IM14 is equipped with a total of **10 sends** that are divided into 6 mono auxiliaries, AUX1 to AUX6 and two stereo auxiliaries, CUE1 and CUE2. Each of the six mono sends has its own level control and PRE fader switch. The stereo sends have level and pan controls. In difference to the mono sends, these sends are defaulting pre fader. A POST fader switch and a CUT switch are assigned to each stereo send. This default setting can be changed to customers requirements; please ask for details.

The **CHANNEL OUTPUT** defaults to post fader and calibrated mode. Without any additional settings, the output signal of the channel fader is available in parallel at the channel output. A set of switches and an additional level control makes it possible to modify the use of the channel ouput. The FDR ON switch inserts the CH-OUT pot pre the channel output amplifier and allows adjusting the level of the output independently. The maximum gain of this level control is 6 dB. The PRE switch changes the default post fader feed to pre fader and the INP (input) switch feeds the output section directly from the input selector. This function has priority over pre. It makes possible to use the channel output as a ‚clean feed', pre EQ, filters and external processing gear for recording on a multitrack device. The N-1 switch has the highest priority of all switches and sets the channel output into **n-minus mode**. The default master rail for the n-minus system is the AUX6 send; however, any other send or group can be assigned as well. A couple of jumpers on the module make possible to select one of the aux sends as n-minus source bus. In addition, a modification in the master section offers the choice to use audio groups as well. The jumpers make possible to use different masters for particular modules. If there are any special requirements that concern the implementation of the n-minus system, please ask. The TB switch enables the ‚Talkback to Channels' function for the particular channel that allows mixing talkback to the channel output. See the description of the Talkback/Oscillator Module for details. The channel output is electronically balanced and at nominal level. Internal trimming facilities make it possible to set the channel output level differently. The source impedance is below 60 Ohms in the transmission band. Depending on the load resistor, the output can drive levels of up to + 30 dBu. Transformer balanced outputs are possible; please ask.

The **PROCESSING SECTION** of the BC-IM14 broadcast mono input module contains a sweeped high-pass filter, a 4-band equalizer and a switched insert section. Each section has its own bypass switch and operates independently.

The **HIPASS-FILTER** is a ‚maximum flat' filter with a steep rate of 24 dB/oct. The edge frequency can be adjusted from below 30 Hz to 600 Hz.

The **4 BAND EQUALIZER** combines two fully parametric bands with shelving, sweeped, high EQ and a ‚soft-bell' type, sweeped low EQ. All 4 bands have a gain range of +/- 20 dB and use distortion free, pristine sounding RC and ‚Vienna Bridge', active circuits. The two fully parametric mid bands are bell type EQ's with a Q factor control range from 1.5 octaves to a third. The center frequency can be sweeped from 300 Hz to 15 kHz with the MID2 and from 60 Hz to 3 kHz with the MID1 band. The shelving high EQ uses a soft RC filter with a steep rate of 6 dB/oct.. The sweep range is from 500 Hz to 25 kHz. The internal circuitry includes constant an upper limit frequency for gain above 40 kHz to assure appropriate rejection of RF noise. The low EQ is always a very soft bell type filter that avoids that subsonic interferences are accentuated with high gain settings. The sweep range is from 20 Hz to 1 kHz.

The switched **INSERT section** is electronically balanced and operates at nominal level. The insert output is always available on the connector panel, while the insert input is switched into the signal chain, when the INS button is pressed. Both, input and output, can handle levels of up to + 30 dBu. The default position of the insert is post equalizer. The PRE EQ switch routs the insert section pre EQ, post high-pass filter.

The **channel FADER** is a conductive plastic, VCA law fader with 100 mm stroke that offers a maximum gain of 10 dB. Faders with 126.5mm stroke and +15 dB gain are optionally available, please ask. The audio path of the fader section uses a high quality VCA, brand THAT. The scale accuracy is better than 1 dB from + 10 to - 20 dB. The zero point is internally calibrated to pinpoint accuracy. The Channel On switch controls the VCA and an additional relay mute circuit. Both, fader and switch can control the start and red light system in different ways. It is possible to reverse the function of the Channel On switch to CUT, therefore from ‚default off' to ‚default on'. However, this version has to be installed with all input modules of the particular console to maintain proper operation of the different master control functions. The output of the fader drives the pan pot, which is a standard, mono to stereo, panoramic control with 3 dB center attenuation. While the pan pot output always feeds the program master rails, the group routing is directly fed from the fader output. The PAN switch makes possible to use the groups in stereo mode. In this case, the odd numbered groups are assigned to the left channel and the even numbered groups are assigned to the right channel. The main routing section combines eight separate switches for the audio groups and an additional switch for each program master.

Six VCA-GROUPS can be selected in parallel by six switches. A LED indicates that one or more VCA groups are assigned to the particular channel fader. The VCA group master faders are installed in the master section. The VCA grouping controls the VCA level setting and the Channel On function. If the VCA group master is not in ON mode, all channels that are assigned to this master are also in OFF mode. This additional features offers the choice to use any number of group masters as mute group, just by setting the VCA group master fader to the 0 dB position. An additional important feature of the VCA grouping system is that it includes the remote control of the fader start functions.

The **PFL system** of the BC4 broadcast console system is a stereo system that can be operated in different modes. With the BC-IM14, mono module, the default function is ‚pre fader listen'. The input signal of the fader is routed to both the left and right PFL audio busses in parallel when PFL is active. The master status function AFL FLIP changes the PFL system into an AFL/Solo system. With this function, the output of the fader or the output of the pan pots feeds the PFL bus. If the pan pot or the fader output is used for AFL is determined by jumpers on the module. The AFL SAFE switch disconnects the particular channel from the master status bus. There are two main PFL modes, add and single. In adding mode, any number of channels are mixed into the PFL bus. In single mode, only one PFL can be active at a time. If PFL is active in a particular channel, the activation of another PFL automatically resets the PFL that was previously active. In addition, a central PFL Reset switch clears all PFL's. The local status of PFL can be controlled by the fader position and/or the channel on switch. The default setting is, that PFL is in latch mode, as long as the channel is off. When the channel is on, the fader position determines if PFL is in latch mode or not. Opening the fader or switching on the channel while the fader position is above threshold resets PFL automatically. With channel on and fader position above threshold, it is still possible to activate PFL by pressing and holding the PFL button. Each of these functions has a corresponding jumper that makes it possible to modify the behavior of the PFL system in any desired way.

The extensive **START and red light control** uses separate interfaces that are automatically selected with the input source selection of the module. Line activates the remote control interface while mic activates the red light control port. With OSC, both interfaces are disabled. Both interfaces can be controlled by the channel on switch and the fader switch. In addition, the START switch makes it possible to toggle the state of the port. Jumpers on the module's PCB determine If a particular control is enabled or not. The default setting is, that with ‚channel off' the entire interface is disabled. When the channel is on, the interface switches to on mode, when the fader is open and back to stop mode when the fader is closed or the channel is switched off. Besides, it is possible to toggle the on state with the START button. The remote interface that is active with line input select can be used in static or pulsed mode. In static mode, the start relay is closed with fader open and the stop relay is open. Closing the fader toggles the state of both relays. With pulsed mode, start causes the start relay to close for 0.3 seconds and stop causes the stop relay to close for the same period. The time can be modified by the value of a capacitor. The state of the interface is indicated by a LED. An opto coupler input makes possible to operate the START LED as ancillary lamp, alternatively. A second LED, FADER OPEN, indicates, if the fader is above threshold or not.

The microphone / red light interface combines a relay, a 24 V/100 mA source voltage for speaker red light and two control inputs for COUGH and TALKBACK. While the relay and the 24 V voltage are simply switched on with the jumper selected way with fader open and/or channel on, the talkback and cough inputs are activated by external switches that connect these inputs to relay ground. COUGH switches the channel off when active. Talkback routs the microphone preamp output to a console bus, that sums all incoming talkback signals. This function makes possible to use a microphone for talkback from the studio alternatively. The output of this bus is part of the console's listen system that is described in detail with the Talkback/ Oscillator module BC-TBO4. The output of this bus can be routed to different speakers, headphones, and outputs. This function can mute the main mic signal automatically. In addition, it is possible to disable this function, if the channel is in ‚on' state.

In addition to the microphone control interface, there are three ‚FADER OPEN' busses in total. The channels can be assigned to different ‚fader open' busses in microphone and line mode. Each of the ‚fader open' busses can be used to control master functions like **‚ON-AIR'** autocut, autodim, talkback disable and more. This principle offers the choice to arrange any desired operation of the control section, just by jumpering the particular input and master modules.

A peak level meter with 10 LED's and a range from +12 dB to -15 dB displays the level at the fader input. A connector block with six 3-pin XLR connectors and two 9-pin D-sub connectors is assigned to each module slot.

analoge + digitale Tonstudiotechnik Karl Jüngling • Inh. Dipl.-Ing. Gerd Jüngling
www.adt-audio.com • sales@adt-audio.com
Scholtwiese 4-6 • D45966 Gladbeck • Germany • Phone 0049 2043 51061 • Fax 0049 2043 56844