RTS™

Matrix News 2003

innovating the future of communications™

Cronus™ Digital Intercom Matrix

RTS™ Cronus™ is a 32-port digital matrix intercom in 2 RU. Based upon an advanced DSP architecture, Cronus has the ability to link up to 4 units into a single 128 port matrix. Through the use of standard video coaxial cable, the individual Cronus units can be located hundreds of meters apart, and still appear as a single matrix.

When connected as a single matrix, the individual Cronus control remains autonomous and independent at each matrix for the highest reliability.

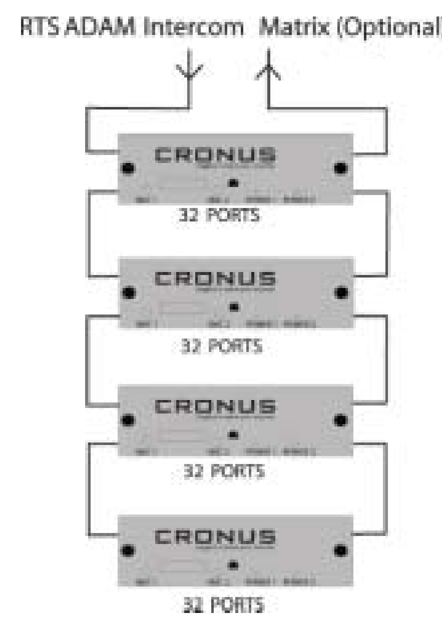
RTS™ Cronus™ "Seamless Sizing" allows 2, 3, or 4 matrices to be combined by the simple addition of coaxial cable. Seamless sizing means that upon interconnection, each matrix and all panels learn about the new configuration without operator intervention or reprogramming. The USB ports for programming are available on both front and rear panels. The front panel display and shaft encoders permit extensive control without the need for a PC.

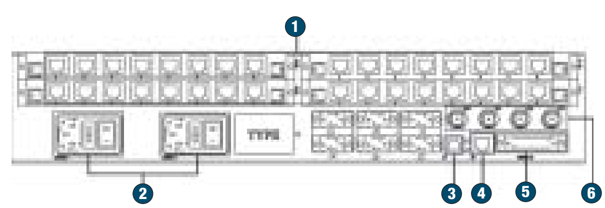
Cronus is FULLY compatible with all existing RTS matrix products. Also, Cronus has the ability to connect, via an interface card, to existing ADAM matrices. This allows Cronus to be a "satellite" matrix to any existing ADAM™ intercom system from a 128 port single frame system to a frame of over 800 ports - a single, large matrix - no trunking, no blocking.

Features:

- **Active Feedback Suppression -** Feedback suppression to allow high volume operation in confined areas without the worry of feedback.
- **Seamless Sizing -** Dynamically resizes the matrices when additions are detected without operator intervention.
- **USB Connectivity -** Convenient front panel access as well as traditional rear access for system programming.
- **Advanced DSP -** Digital signal processing designed to support audio signal processing as well as VOX on all 32 ports (inputs).
- **Modular Architecture -** The modular architecture allows for port expansion from 8 to 32 ports giving easy user expandable systems in the field. Also, users can choose from a variety of intercom cards such as VOIP, AES-3 and Analog, each of which support 8 channels or ports.
- **Redundant Power Supply -** Each Chassis is powered by two internal power supplies, either of which can sufficiently power all the equipment ALONE. The power supplies have separate AC feeds for the ultimate in redundancy and protection.

Seamless Sizing Diagram





1. Intercom Ports
2. Redundant Universal AC Power Supplies
3. USB Programming Connection
4. LAN Programming Connection
5. GPI/O Connections
6. Matrix Expansion Links

J1 AZedit Serial Connection

J2 TBD

J3, J4, J5, J6 – RS 485/422 For UIO/PAP/LCP-102 Panels

INDUSTRIAL MATRIX INTERCOM KEYPANEL

WKP-1

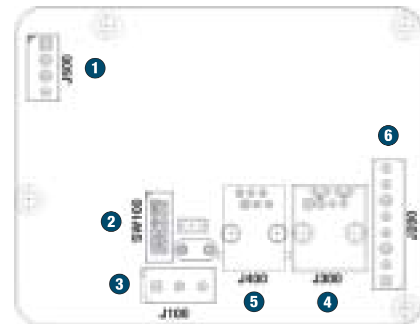
Industrial Matrix Intercom Keypanel

The WKP-1 keypanel is the industrial keypanel from Telex. It offers simplified operation and integrates seamlessly with RTS™ digital intercom systems (RTS™ ADAM™/Zeus™/Cronus intercom systems). The physical size and weather-resistant design provides a flexible and robust intercom system. The GPI relay switch lets the user set up door latching, unlatching and other related actions by pressing a single button from any panel in the system.

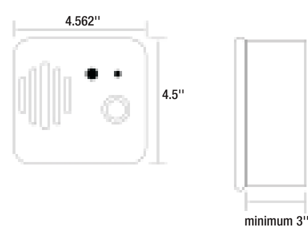
Features:

- **Environmental -** The unit sustains exposure to rain, snow, and direct sun, allowing it to work in all environments. It operates in temperatures ranging from -30° F to 120° F. The mechanical design of the anodized, aluminum front panel and non-removable volume control knob provide a vandal resistant panel.
- **GPI Relay -** The WKP-1 has a local GPI relay four-pin terminal contact closure to give the user the functionality of latching, unlatching doors or other related functions, with the press of a button.
- **Front Panel -** Single button interface for push to talk operation. There is a built-in waterproof speaker for listening capabilities. The panel has three LEDs used to indicate power ON, microphone ON, and Incoming Call Signaling.
- **Microphone -** The WKP-1 keypanel has a built-in waterproof Electret microphone. The microphone can be "hot mic" enabled or disabled through the use of a dip switch setting.
- **Key Panel Assignment -** Key assignments, GPIO operation, and panel listen/muting functions are set via AZedit™ Software for the intercom matrix.
- **Audio Control -** Vandal and weather resistant front panel volume controls adjusts listen level to the internal speaker. An internal limit adjustment sets the lower level of minimum volume achievable.
- **Audio Processing -** An integral limiter/compressor insures maximum intelligibility of the spoken word. The compressor uses the RTS Standard compression design.
- **Dual Power -** The keypanel can be powered locally or remotely via a CAT 5 connection from the matrix.

1. GPI Relay Contact 4-pin Terminal
2. Local or Remote Power Switch
3. Local Power 3-Pin Terminal
4. RJ45/CAT-5 Connection
5. 8-pin Terminal
6. RJ12 Connection



The keypanel fits a standard 2-gang, minimum 3" deep electrical box*.
Electrical box not included with keypanel



Terminal Strip	RJ45	RJ12
RS485+	Audio_IN+	Data-
RS485-	Audio_IN-	Audio from Matrix+
NC GND	Audio to Matrix+	
Audio_OUT+	RS485+	Audio to Matrix-
Audio_OUT-	RS485-	Audio from Matrix-
NC Power	Data+	
Audio_IN+	Audio_OUT+	N/A
Audio_IN-	Audio_OUT-	N/A

WKP-1 Industrial Matrix Intercom Keypanel Specifications

Matrix

Input/Output:	8 dBu nominal,16 dBu maximum

Panel Mic Input

Mic Type:	Electret Condenser
Nominal Level:	-44 dB ±3 dBu
Directivity:	Omni Directional

Speaker

Output:	3 watts

Power Requirements

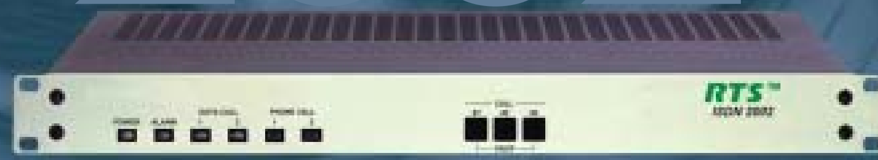
Power Requirements:	15 VDC 250 mA for local power 24 VDC up to 500 mA for remote power "300 or more"

Connections

- RJ45 for use with Standard CAT-5 cabling
- RJ12 for use with RTS Matrix Standard 3-pair cabling
- Terminal Strip for use with unterminated cabling

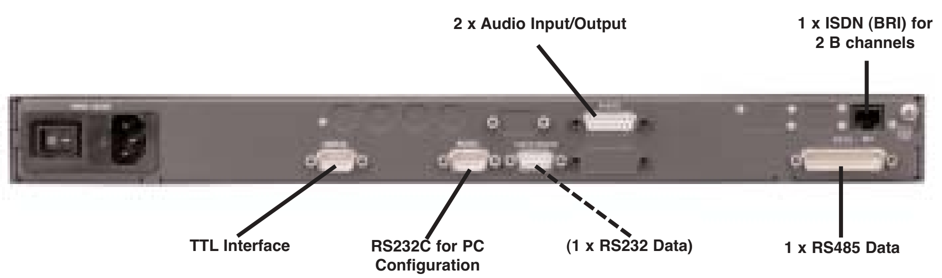
ISDN 2002

ISDN TELEPHONE INTERFACE



General

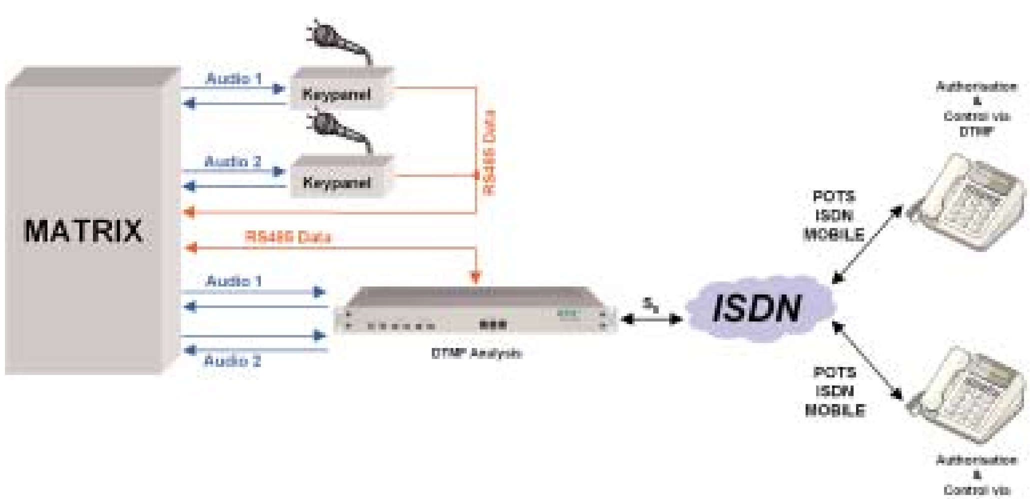
The RTS ISDN 2002 System incoporates two Audio codecs according to G.711 and G.722 coding algorithms for bidirectional communication.The coding delay of the 7-kHz G.722 coding algorithm is less than 10ms. The two Audio codecs included in the system can be configured as 3.1-kHz (G.711) or 7-kHz (G.722) Audio codecs. Each coded Audio signal occupies a single ISDN B channel (64-kbit/s channel). Therefore both codecs are using 2 B channels of the So interface. In the 7-kHz coding mode also a data signal can be transmitted. Two different types of data interfaces are available. In the G.711 Telephone mode the system detects the DTMF tones received from a standard Telephone set and converts the tones into the Telex protocol for the control of Telex keypanels. The system can be configured as 2x G.711 Audio codecs with two separate Audio interfaces but one common RS485 data interface. In the 7-kHz (G.722) Audio codec mode a RS232 data channel is available. This data channel can be used for the interconnection of a Matrix to a Trunkmaster system. If both Audio codecs are configured as G.722 codecs also two RS232 channels are provided.



Operating Modes

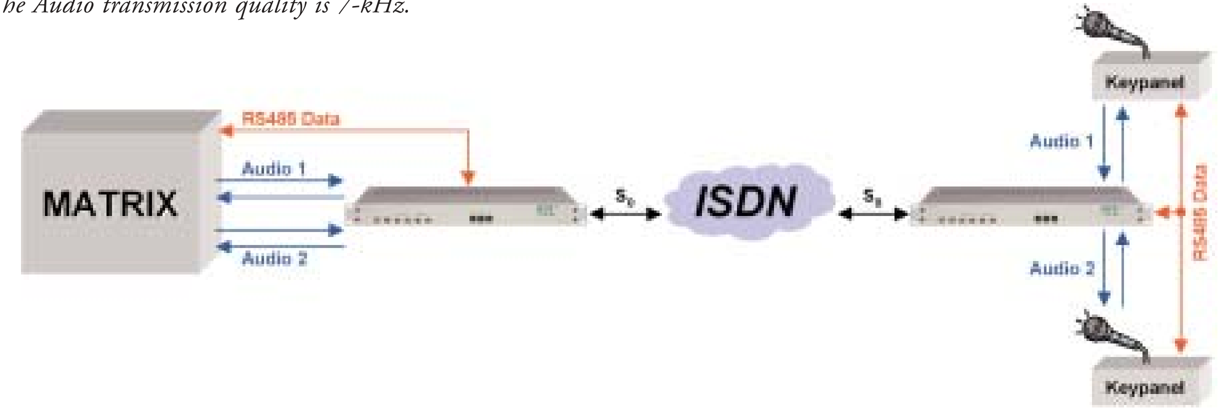
Model 1: Telephone to keypanel

In this operating mode two Telephone sets can dial a RTS ISDN 2002 system. Via the DTMF tones of the telephones the keypanels can remotely be controlled. The system converts the received DTMF tones into the Telex protocol.



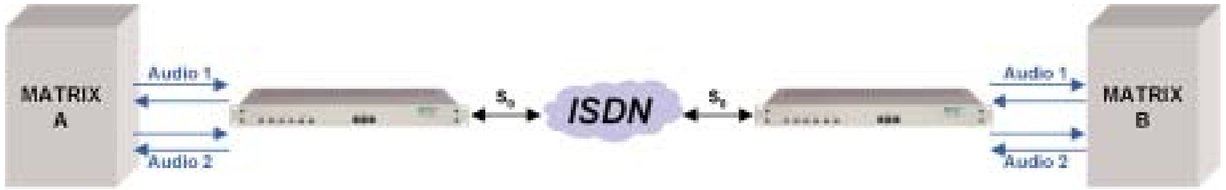
Model 2: Keypanel to Matrix

At both sides RTS ISDN 2002 products are installed. The RS485 commands of the keypanels are transmitted to the Matrix for routing purpose. The Audio transmission quality is 7-kHz.



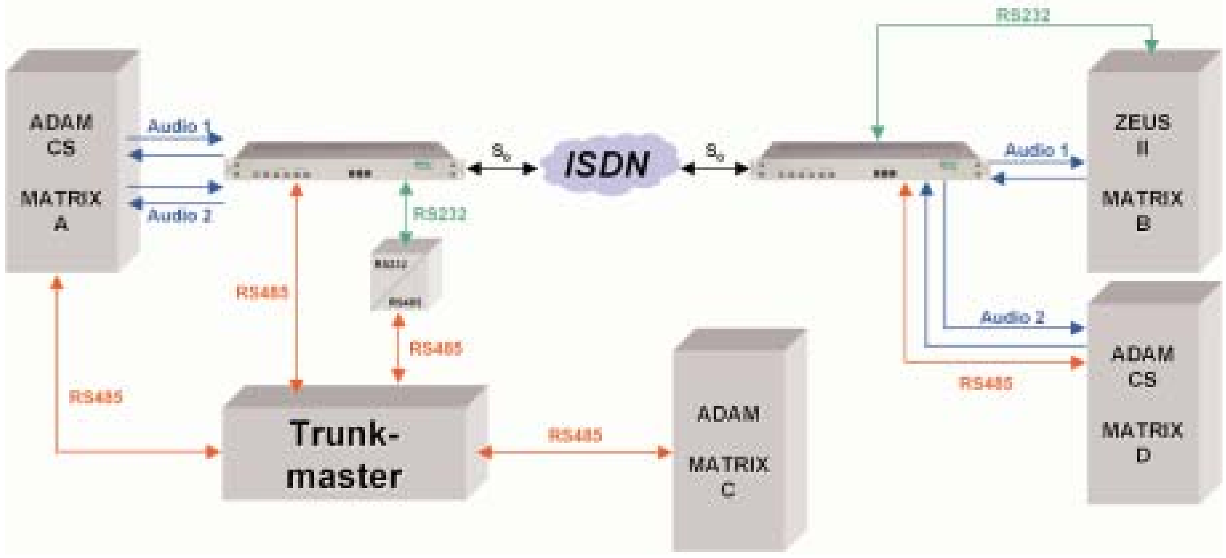
Model 3: Matrix to Matrix

Matrix units are interconnected by 7-kHz transmission channels. Only Audio signals are transmitted.



Model 4: Matrix to Matrix with Trunkmaster

In addition to Mode 3 also a RS232 Data channel from the remote Matrix can be transmitted to the Trunkmaster.

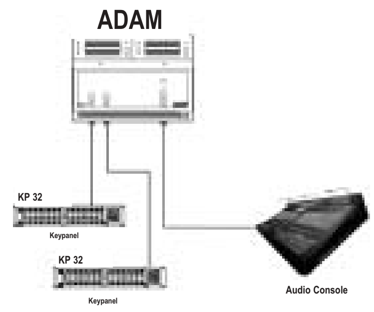


AES-3

DIGITAL AUDIO INTERFACE

Digital Audio Interface Cards

The AES-3 Digital Audio Interface Card expands the connectivity to the ADAM® Intercom by supporting AES-3 over coaxial cable. It supports 8 audio channels in and out through 8 AES-3 connections. The AES-3 card provides connectivity to any other third party AES-3 audio device. It supports all standard hot swappable and configurable features within the ADAM Intercom family. This card supports incoming sample rates from 16 kHz to 108 kHz with 24 bit audio. Outputs are compatible with all AES-3 recommended procedures. Along with the newly introduced RVON-8 VOIP card, the RTS ADAM intercom systems now natively support analog, AES-3 digital and VOIP signal formats.



Features:

- The AES-3 cards are hot swappable to allow live insertion into any IO slot in the ADAM frame.
- The TDM and Control Bus interfaces are compatible with the current and next generation ADAM matrix TDM and control buses.
- Hardware and firmware compatible with current and next generation ADAM matrix systems.
- Provides downloadable firmware features through AZedit.

AES-3 Specifications

Performance Specifications:

Dynamic Range:	120 dB
Input Sample Rate (serial input port - Fsi):	16 to 108 kHz
Output Sample Rate (Fso):	44.1 kHz
Output Data:	16 bits capable of 24 bits
Output to Input Sample Ratio Rate:	.33 to 3
Total Harmonic Distortion + Noise:	120 dB
Peak Idle Channel Noise Component:	-140 dBFS
Resolution:	16-24 bits

Digital Filter Characteristics:

Passband:	
Upsampling:	.4535 Fsi Hz
Downsampling:	.4595 Fso Hz
Passband Ripple:	±0.007 dB
Stopband (downsampling):	.5465 Fso Fsi/2 Hz
Stopband Attenuation:	110 dB
Group Delay:	175 ms
Group Delay Variation vs. Frequency:	0.0 µs
Interchannel Phase Deviation:	0.0°
High-Level Input Voltage, except RXP, RXN:	2.0 - (VD+)+0.3 V
Low-Level Input Voltage, except RXP, RXN:	-0.3 - 0.8 V
Low-Level Output Voltage (Io=-20 uA), except TXP, TXN:	0.4 V
High-Level Output Voltage (Io=-20 uA), except TXP, TXN:	(VD+) -1 V
Input Leakage Current:	±10 - ±15 µA
Differential Input Voltage, RXP to RXN:	200 mV
Output High Voltage, TXP, TXN:	(VD+) -.07 (VD+) -0.4 V
Output Low Voltage, TXP, TXN:	0.4 - 0.7 V

AES-3 Specifications *continued*

Switching Characteristics:

RST Pin Low Pulse Width:	200 ms
OMCK:	
Frequency for OMCK = 512 Fso:	4.096-55.3 MHz
Low and High Width for OMCK = 512 Fso:	8.2 ns
Frequency for OMCK = 384 Fso:	3.072 to 41.5 MHz
Low and High Width for OMCK = 384 Fso:	12.3 ns
Frequency for OMCK = 256 Fso:	2.048 - 27.7 MHz
Low and High Width for OMCK = 256 Fso:	16.4 ns
PLL Clock Recovery Sample Rate Range:	8.0 - 108.0 kHz
RMCK Output Jitter:	200 ps RMS
RMCK Output Duty Cycle:	40-60% (typ. 50%)
RMCK Input Frequency:	2.048 - 27.7 MHz
RMCK Input Low and High Width:	16.4 ns
AES-3 Transmitter Output Jitter:	1 ns

RVON-8

VOICE OVER NETWORK CARD

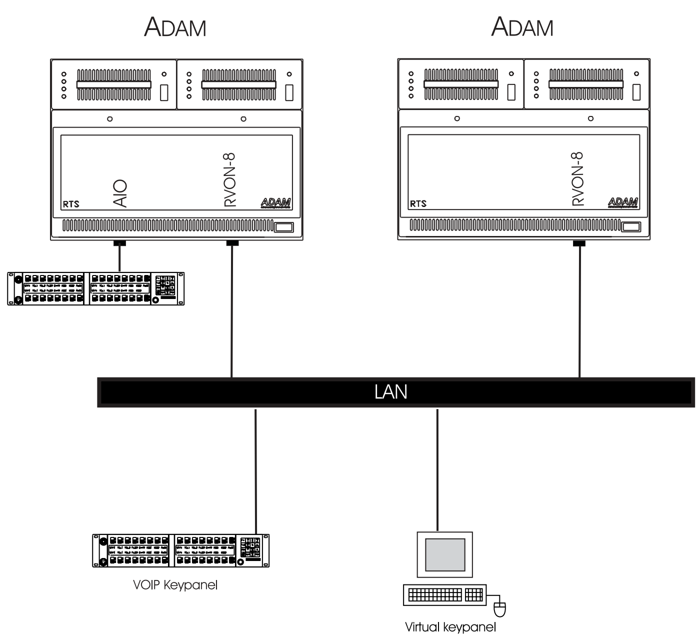
RTS™ Voice Over Network Card

Installed directly into ADAM™ Matrix Intercom Systems, the RVON-8 (RTS Voice Over Network) Card expands the connectivity to the ADAM Intercom by supporting 8 audio channels (ports) in and out. Each channel has configurable network and bandwidth parameters that can be tailored to individual network functions, as well as ancillary data for keypanels and trunking control. As with all ADAM Intercom family products, the RVON-8 card supports all standard hot swappable and configurable features through AZedit™ configuration software. The RVON-8 card supports both Telex Intelligent Trunking over IP, as well as support for remote keypanels and virtual keypanels via VOIP. Fully compatible with Internationally recognized standards, the RVON-8 card supports the following protocols: G.711, G.729AB, G.723(2 speeds), and G.722.

Features

- **Installation -** The RVON-8 card is hot-swappable and installs in any available slot in an ADAM Intercom System. The card provides a single RJ-45 Ethernet connection for use with any 10/100 base network.
- **Trunk Capable -** The RVON-8 card supports ancillary data control for use with Telex® Intelligent Trunking.
- **Ethernet Compatible -** Fully ethernet capable. RVON-8 uses standard ethernet protocols and is compatible with all Ethernet compliant devices and networks.
- **AZedit Configurations -** The user has the ability to adjust the audio quality of the matrix intercom to the available bandwidth on the network.
- **Addressing -** Eight individually addressable audio channels. The RVON-8 card can feed simultaneously VOIP (voice over internet protocol) capable key panels as well as various other matrix intercom systems.

System Diagram



RVON-8 Specifications

Digital

Compression:	Bit Rate (/sec):	Coding Delay:
G.711:	64 K	125 µs
G.723:	5.3 K/6.3 K	30 ms
G.729 AB:	8 K	10 ms

*Data Rate Depends on Codec Selection

Connections

- RJ-45 Ethernet via backcard
- DB-9 Serial Port via backcard
- High-Density keyed ADAM Compatible Backplane Connector

Power:	12.5 w / 2.5 A at 5 V
Physical:	5.687" W x 11.024" L

4WBA

F O U R   W I R E   B U F F E R   A M P L I F I E R

4WBA Four Wire Buffer Amplifier

The 4-wire Buffer Amplifier is a six channel, high performance audio line driver amplifier with transformer balanced inputs and outputs. It can be used as a six input, six output group of individual amplifiers in a 1 RU mount. Electrical isolation between amplifiers is sufficient to eliminate any chance of crosstalk. Each channel on the 4WBA system has its own level control, in/output level monitoring 3-point LED meter, limiter, gate and VOX tally output function for RX channel.

Power Requirement:	Single Output Power Supply +12 V 2.5 A/30 W (2 PS unit for Redundant) DC converter ±12 V 0.45 A (on Board)	
AC Inputs:	AC 100 V to 240 V ±10% Input 2 Fused AC Power Connector with Line Filter	
Input Level Adjustment Range:	+30 dB to -10 dB and 6 step	
Only RX-Ch:	<u>GAIN Range</u>	<u>VR Adjustable Gain</u>
	+30 dB	+20 to +40 dB
	+20 dB	+10 to +30 dB
	+10 dB	0 to +20 dB
	0 dB	-10 to +10 dB
	-10 dB	-20 to 0 dB
Adjustable VR ranges +/- 10dB with center point for Knob Range		
Maximum/Minimum Input Level:	+20 dB to -40 dB	
System Frequency Response:	TXch: 50 Hz~10 kHz (-3 dB) RXch: 80 Hz~10 kHz (-3 dB)	
Noise Level:	> -60 dB (Bandwidth 30 kHz)	
Distortion:	>0.5% (under limiter activate point)	
Frequency Range:	100 Hz to 10 kHz (0 dBu Output)	
Crosstalk:	> -60 dB, at 30 Hz~10 kHz > -80 dB, at 1 kHz	
CMRR:	> -70 dB, at 30 Hz~1 kHz > -50 dB, at 15 kHz	
Input Circuit Method:	RXch - Trans Coupling TXch - Electric Balance	
* Transformer Coupling is Option ** Floating/Unfloating are selected by jumper. *** Only RXch 600 W Termination resister is selectable by jumper.		
Maximum Output Level:	Operating Level is selectable 0 dBu and -10 dBu by jumper. Selected in a jumper: In case of adjustment 0 dBu for operating level RX 600 W load Maximum +10 dBu (1 kHz) TX 600 W load Maximum +10 dBu (1 kHz) In case of -10 dBu for operating level, Limiter is changed by 0 dBu	
Output Circuit Method:	RXch - Electric Balance TXch - Trans Coupling	
***TX system Floating/Unfloating are selected by jumper.		

Features

- **Transformer Balanced Input -** Each input uses two cross-coupled audio transformers to help minimize hum and provide galvanic isolation.
- **Impedance -** Input impedance selection to either 600 Ohms or 10.0 K Ohms via 3 pin jumpers.
- **Gain -** -40 to +20 dBu, five step selectable and ±10 dB gain volume for RX channel operating level. -10 to +10 dBu adjustable for TX channel operating level. RX channel is available for input mute position. Nominal setup level is selectable 0 dBu or -10 dBu, via jumpers
- **Limiter -** Threshold level is adjustable -6 to +12 dBu, via inside trimmer.
- **Gate and VOX -** Threshold level is adjustable -30 to -10 dBu, via panel trimmer. Each RX channel of VOX tally output DB15F connector for the UIO256 GPI function.
- **Transformer Balanced Outputs -** Four Wire side of In/Outputs capable of driving 600 Ohm loads. Matrix site In/Output stages transformer option.
- **Redundant Power Supply -** Dual power supply provides redundant operation. Bi-color LED display and Fault LED with tarry output to DB15F connector.

4WBA Specifications *continued*

Audio Gate Activate Level:	-30 dbu~-10 dBu Adjustable Pot on Front
Audio Gate Attack Time:	General audio 30 mSec nominal
Audio Tally Output:	@12 V 50 mA load operate is possible. An output is negative logic Mute = High VOX = Low
Equipment for Audio Limiter Circuit:	Limiter activate level ranges -6 dB to +12 dB. It adjusts by internal pot.
Adjustment Value:	+10 dBu
Signal Level Indicators:	Display output level to all Ch. >+10 dBu Red LED Turn ON > 0 dBu Yellow LED Turn ON > -20 dBu Green LED Turn ON
Equipment for Audio Filter:	Only audio gated ch., High Pass filter at 80 Hz, Low Pass filter at 10 kHz

Telex EVI Audio GmbH
Hirschberger Ring 45
94315 Straubing - Germany
telefon + 49 (0) 9421 706 - 0
fax: + 49 (0) 9421 706 - 265
www.rtssystems.com