MATRIX TW

RTS™


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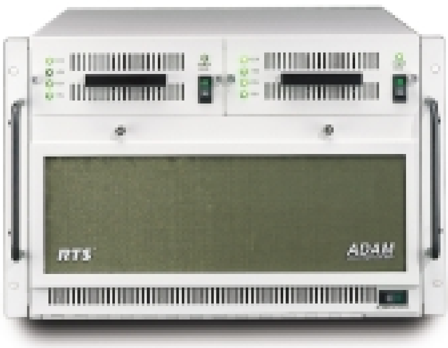
Contents

ADAM®

Advanced Digital Audio Matrix



ADAM®

ADAM CS®

The RTS ADAM and Zeus range of digital intercom matrices offer a flexible solution to any customer's intercom requirements.
All three matrices within the family are fully compatible, sharing the same electronic design, software control system and range of keypanels and accessories.
The RTS ADAM intercom system is a state of the art intercom system for medium to large installations. It provides CD quality audio for all signals, and will support any combination of up to 136 keypanels or 4 wire ports within a single 7 rack unit frame. System sizes as large as 1,000 ports are also possible. The system offers full time, full access communications between all users without limitations.
The ADAM matrix is also expandable beyond 136 users with the addition of the bus expander card. This module plugs into the standard ADAM chassis and allows for the digital audio bus to be extended to a second frame. The interconnection required is simply two coaxial cables. This concept maybe expanded to a third or fourth frame by the addition of further bus expander cards and coaxial cables. It should also be noted that the individual frames maybe located up to 300 meters apart using this technique.
The ADAM matrix uses an AIO interface card as its audio 'engine'. This card interfaces and processes the analogue audio inputs and outputs to the internal digital TDM bus. Each card will support up to 8 users. The ADAM frame maybe populated with up to 17 of these cards. Cabling to either keypanels or 4 wire audio connections is achieved with either RJ-11 or DB-9 connectors. A third interconnect using RJ-45 connectors is also available. This allows the installer to easily separate the audio signals from the data thus simplifying jackfield installation.
As standard, ADAM ports are electronically balanced for both input and output. Transformer isolated rear connector units are available as an option.
The ADAM frame comes standard with dual redundant, auto-switching power supplies and system controllers.
ADAM also supports a number of third party solutions for the following standards:

- Telephone (2 wire analogue)
- Fibre
- ISDN
- E-1
- LAN / WAN

The ADAM CS is ADAM's 'baby brother', providing a 64 port matrix in just 5 rack units. The unit is expandable from 8 to 64 by 64. In addition there are two further slots for master and slave system controllers. As standard the ADAM CS is sold with a single controller card and dual power supply units.
The ADAM CS is available in three different rear connector configurations, including DB-9, RJ-11 and 25 pair U.S. Telco style connector.
As with the ADAM, the CS Controller cards contain flash memory, this means that all set up and configuration information will be retained in the event of a power failure. This is particularly important when the system is installed in an outside broadcast vehicle
The Zeus is a 24 user, fully programmable digital intercom. The matrix is housed in two rack units and offers the following features as standard:

- 24 ports (balanced analogue audio input / output and RS 485 data).
- A to D conversion, 44.1 kHz sampling 20 bit resolution.
- Audio input and output level control.
- Crosspoint level control.
- GPI port. 8 input and 8 output.
- RS232 port for ZeusEdit - PC interconnect.

Zeus

Digital Audio Matrix

ZEUS®

The Zeus is a 24 user, fully programmable digital intercom. The matrix is housed in two rack units and offers the following features as standard:

- 24 ports (balanced analogue audio input / output and RS 485 data).
- A to D conversion, 44.1 kHz sampling 20 bit resolution.
- Audio input and output level control.
- Crosspoint level control.
- GPI port. 8 input and 8 output.
- RS232 port for ZeusEdit - PC interconnect.

The Zeus comes complete with ZeusEdit software. This is a full feature configuration software running on a PC. ZeusEdit allows the user to edit or programme any of the matrix features either on or off line. Configurations maybe stored on the PC, recalled and downloaded to the matrix at any time.

AdamEdit

All RTS matrices are supplied complete with configuration software. AdamEdit is supplied with both the ADAM and ADAM CS whilst ZeusEdit is supplied with Zeus. In all cases the configuration software runs on a standard PC under Windows and is compatible with versions 3.1, 3.11, 95, 98 and NT. The software can be operated either ‘on’ or ‘off line’. In the On Line mode the user is provided with the exact system parameters as the software uploads the current matrix configuration. This gives the user a high level of confidence as he can be sure that when any edits are made they will not over write or conflict with the current setup. Off line configurations or maps maybe created at leisure, allowing the programmer time to build up from an existing configuration or starting from scratch.
AdamEdit represents the most powerful on line programming tool available for broadcast intercoms. The user may programme or edit any of the following parameters:

- Communication types.
- Talk, listen, talk and listen.
- Point to Point, Group or Party line (Conference) IFB & Isolate.
- All functions are user assignable using 4 character alpha numeric.
- Control.
- Central GPI, up to 255 inputs & outputs available.
- Local panel GPI, 4 inputs and outputs per keypanel (KP 32 and KP 12 only)
- User Programmable Logic (UPL) build simple or complex logic statements to satisfy any communications requirement.
- Audio.
- Force / inhibit crosspoints directly.
- Inspect crosspoint. Status, audio level & controlling parameter.
- Adjust input and output audio levels.
- Inter panel dimming to prevent acoustic feedback.

Technical data	ADAM™	ADAM-CS	ZEUS
Matrix type	TDM Bus	TDM Bus	TDM Bus
Signal-Format	44,1 kHz Sampling, 16 Bit Datas	44,1 kHz Sampling, 16 Bit Datas	44,1 kHz Sampling, 20 Bit Datas
max. System capacity	1000 Ports	64 Ports	24 Ports
Crosspoint Control	Variable Gain Individual Crosspoint Level Control		
Compatibility	CS9000 Series/ZEUS/ADAM-CS	CS9000 Series/ZEUS/ADAM	CS9000 Series/ADAM/ADAM-CS
Accessories	CS9000 Series/ZEUS/ADAM/ADAM-CS	CS9000 Series/ZEUS/ADAM/ADAM-CS	CS9000 Series/ZEUS/ADAM/ADAM-CS
Port Connection	DB-9, RJ-11	DB-9, RJ-11	DB-9, RJ-11
Analog In- and Outputs			
Nominal Level	6dBu	6dBu	6dBu
Maximal Level	20dBu	20dBu	20dBu
Frequency response	15 Hz - 19,5 kHz ± 1dB	15 Hz - 19,5 kHz ± 1dB	20 Hz - 20 kHz ±1dB
Signal to Noise Ratio	> 75 dB unter 20 dBu, A-bewertet	> 75 dB	> 80 dB
Signal to Crosstalk Ratio	< 0,05 % (bei 20 dBu)	< 0,05 % (bei 20 dBu)	0,01 % (bei 6 dBu)
Overtalk	> 75 dB	> 75 dB	-80 dB
Power	100-250 V AC 50 / 60 Hz	90-250 V AC 50 / 60 Hz	90-250 V AC 50 / 60 Hz
Power connector	IEC 320	IEC 320	IEC 320
Size	7 HE x 19“ x 408 mm	5 HE x 19“ x 508 mm	2 HE x 19“ x 406 mm
Weight	25,85 kg		

new KP-32

KP 32

The RTS™ Model KP-32 Keypanel is two rack spaces high and has 32 lever keys: 30 keys for intercom talk/listen assignment; 1 key for call waiting respond/clear; and 1 key for headset/microphone switching. The KP-32 combines all of the programmable features of the KP9X Series keypanels and the KP-12 keypanel. Plus, it adds significant new features such as digital signal processing and binaural headset operation with left/right assignment of audio signals. The KP-32 also introduces large, super-bright, long-life flourescent displays with adjustable brightness control, making it suitable for all types of ambient lighting from direct sunlight to darkness.

Features

- Super-bright, flourescent displays. Provide much better visibility and useable life LCD displays. A display saver mode with programmable scrolling message extends display life and conserves power during periods of inactivity.
- 32 lever keys, with 30 keys available for full Talk/Listen configuration. Doubles the number of channels over the KP9X Series keypanels. Keys support both latching (hands-free) and momentary (push-to-talk) operation.
- Enhanced programming keypad. Provides the complete KP9X keypad command subset, plus new keypad commands and an extensive scrollable menu system. Menus include helpful prompts to walk the user through setup.
- Only 90 mm deep behind the front pannel (approx. 130 mm with connectors). Perfect for consoles, OB vans, etc.
- Binaural (5-pin) Headset Connector: Works with the DSP mixing feature. Lets you independently assign intercom, mic, and programm audio to left or right headphone. Note: monaural (4-pin) connector available as an option. For monaural operation, the mixer lets you select which items are monitored in the headphones.
- Upgradeable. Firmware updates can be received via the internet, for example, and then downloaded to the KP-32 via the intercom connection. Ready for future communication enhancements, including coax, fiber, and ISDN.

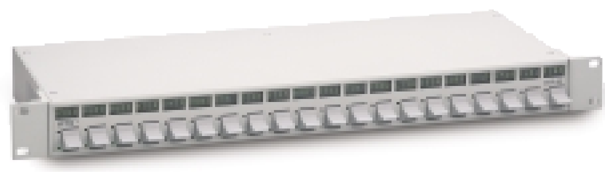
Microphone Preamplifier	
Audio Input Level (at 1kHz)	
Electret Mic:	-42 dB, 5 kOhm
Dynamic Mic	-60 dBm, 150 ohms
Output Level (to matrix)	+6 dBu, ± 0.2 dBu
Max Voltage Gain, Mic to Line	70 dB, ± 2dB
Frequency Response	100 Hz to 800 Hz, ± 3 dB
Test Oscillator	
Output Level (to matrix)	+6 dBu, ± 2 dBu
Output Frequency	400 Hz
Headphone Amplifier	
Maximum Voltage Gain	30 dB
Frequency Response	100 Hz to 8000 Hz, ± 3 dB
Headphone Impedance	50 to 600 Ohms
Output Power	150 mW into 50 ohms
Output Voltage Level	8 volts peak-to-peak (max.)
Mute	Full: 70 dB min. Partial: 0 to 30 dB
Sidetone Range	30 dB

Speaker amplifier and Speaker	
Frequency Response	100 Hz to 8000 Hz, ± 3 dB
Output Power (per amplifier)	5 Watts into 4 ohms
Output Voltage Level	16 volts peak-to-peak (max.)
Volume Control Range	70 dB
Mute	Full: 70 dB min. Partial: 6 dB, to 30 dB adjustable
Speaker	2 watts continuous, 4 watts peak
Intercom Line Input/Output	
Output	+ 6 dBu, ± 0.2 dBu nominal
Input	Nominal: + 8 dBu Peak: ± 20 dBu max.
External Line Input	+ 6 dBu nominal
General	
AC Supply	External, switching type, 90-264 VAC, 50/60 Hz with locking DIN connector for attachment to the keypanel and universal IEC connector for connection to various AC main cords
Environmental	
Storage	-40°C to + 60°C
Operating	-10°C to + 41°C

The KP-12 family of key panels has been designed for operators who need a flexible communications interface but have limited rack space. The KP-12 keypanel is one rack unit high and only 165mm deep. It features 12 lever keys, or push buttons each with a 4 character alpha numeric display. The gooseneck microphone is a plug in type and there is a built in loudspeaker. The 5 pin headset connector will interconnect with binaural headsets. To enable local panel programming the panel is fitted with a shaft encoder and 8 character display. This allows the operator to reprogram keys, initiate telephone dialling and access to the panel setup menu.In order to increase the number of keys, expansion key panels maybe used.

KP 12

The EKP 20 features 20 lever keys, or push buttons each with a four character alpha numeric display. This unit interconnects to the KP 12 via a 8 pin connector, whilst the interconnect to the matrix remains as standard. With expansion key panels it is possible to have up to 64 keys at a single operator position (i.e. one matrix 'port')



EKP 20

As with the KP 32 it is also possible to have crosspoint level control within the KP 12 family. This is achieved using the **LCP 12** or 20. As with the EKP20, this unit interconnects with the KP 12 via an 8 pin connector. Used in combination with the KP 12, the LCP12 will allow the operator to adjust the level of the sources associated with the 12 keys on the KP 12 - assuming that they are programmed. The LCP 12 is capable of adjusting the crosspoint level from +6dB to 'off '. The LCP 20 performs the same operation but in combination with the EKP 20.

LCP 12

KP 96 family of panels represents the most widely installed keypanel type in the RTS range. Originally designed to operate with the CS9500 analogue intercom matrix, the KP 96 panels are fully compatible with ADAM and Zeus matrices. The key panel features 15 programmable talk lever keys each with a 4 character alpha numeric display. Associated with each talk key there is a programmable push button for the listen function. The gooseneck microphone is fixed and the associated loudspeaker is mounted behind the keypad. A headset connector is provided to allow for binaural headset operation. The keypad allows the operator to reprogramme the keypanel and access telephone dialling via the matrix using the TIF 951 telephone hybrid.
As with the KP 12, the KP 96 maybe expanded by adding the EKP 96, this will allow for a further 16 talk lever keys, each with an associated 4 character alpha numeric display, and the associated listen push button. Up to 3 EKP 96 maybe added to a KP 96, this is achieved using a 25 pin cable assembly.

KP 96

In order to access crosspoint level control the **LCP 100** is required. This key panel also interconnects to the KP 96 and will allow the operator to adjust the crosspoint level from +6dB to mute. One LCP 100 maybe connected to each KP 96 and EKP 96 providing level adjustment for all listen keys.

LCP 100

DKP 12

DKP 8

TKP 4

BKP 4

Developed from the KP 12, the **DKP 8 & 12** provide the user with a matrix keypanel in a compact desk top package. Both units feature a plug in gooseneck microphone, built in loudspeaker and front mounted binaural headset connector. The units have either 8 or 12 lever keys or pushbuttons. Each key has a four character alpha numeric display. In order to set up and reassign the desk top key panel, the units are also fitted with a shaft encoder and 8 character alpha numeric display. This allows the user to reprogramme the unit on line. Both units interconnect to the matrix via RJ-11 or DB-9 connectors using a 3 pair cable, and are mains powered.

In order to respond to the growing need for low cost keypanels, particularly when they are used in conjunction with Zeus, RTS has developed a range of easy to use, low cost keypanels.

TKP4, BKP4 and MKP4 have the same feature set but provide the user with three different form factors, TKP will rack mount adjacent to a Tektronix waveform monitor, BKP is a desk top unit and MKP is a single rack mount. Each unit features 4 lever keys with talk and listen operation, in addition there is a call waiting key with a four character alpha display. This allows the user to respond to any incoming caller to the keypanel. It is also possible to reconfigure or change the complete set up of the keypanel using the menu keys. These keys permit copying of the reply key function to any other key and also allow the user to access the complete directory for ports, party lines, IFB's etc.

All panels are mains powered and option DB-9 or RJ-11 connectors for the matrix interconnect. TKP4 and BKP4 have built in loudspeakers, whilst the MKP4 maybe used with the MCS325 Modular Speaker from the RTS TW range. All three units feature plug in microphones.

MATRIX

The SSA424 Dual Digital Hybrid interfaces two 2-wire intercom lines to a digital matrix intercom. The 2-wire circuits maybe either balanced or unbalanced. This allows the user to interface either RTS TW or Audiocom systems to a matrix intercom. The SSA424 features advanced digital signal processing such that the null between the 2-wire and 4-wire circuits is achieved automatically.
The front panel of the unit has peak reading meters which allows the user to 'match' the 2-wire audio level to the 4-wire audio level.
The SSA424 makes the 2/4 wire interface simple with little or no 'setup' required. This means the user can interconnect the two systems and 'add' users to the 2-wire system without any need to reset or adjust the interface.

In addition to the audio interfacing it is also possible to integrate the call signal from the two wire system with the matrix intercom. This is achieved by the addition of the call signal option board. This device provides a contact closure when the call tally is 'on' in the 2-wire system. This maybe used to control a GPI in the matrix intercom to turn crosspoints on or off.

The TIF951 Digital Telephone hybrid is a two line analogue telephone interface for the matrix intercom. The unit provides bi-directional communication between the matrix and analogue telephone lines and will pass and detect DTMF dialling tones. This allows a user within the intercom to dial into the phone system from his standard intercom keypanel. Telephone numbers maybe either stored and recalled or dialled directly using the keypad on the intercom panel. External users may access any user within the intercom by calling the TIF and then dialling a DTMF code which corresponds to the directory number they wish to communicate within the intercom. By this means the matrix and telephone system appear seamless to the user and multiple conversations are easily managed. It should be noted that an external caller may access any type of communication including point to point, group, party line or IFB.

The UIO256 is a one rack unit interface for GPI inputs and outputs to any matrix intercom. The unit provides 16 GPI inputs and outputs with a front panel LED tally. This unit will interconnect with any of the matrix intercoms using a 2 pair cable. Up to 16 x UIO256 maybe connected to a single matrix frame, providing 255 GPI inputs and outputs. All GPI's are programmed within ADAMedit. Once programmed a GPI can be made to perform any action within the system which can be programmed from a keypanel.

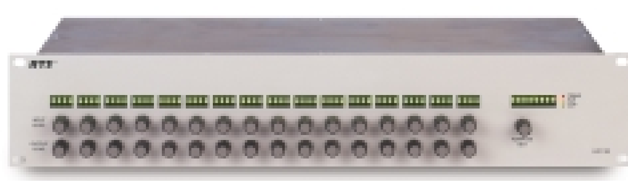
The LCP102 is a two rack unit panel designed to adjust input and output audio level at the matrix. The unit features 16 sets of controls which comprise 2 shaft encoders and a 4 character alpha numeric display. In addition there is an 8 character display with a shaft encoder for menu options.
The unit will adjust input and output level for any of the matrix audio ports. Each group maybe programmed and the display will show the port alpha. Any adjustments which are made will also be reported in ADAMedit and as they are made the 4 character display will show the relative level in dB's. The LCP 102 is particularly useful in Master Control Rooms or Outside Broadcast vehicles where incoming lines are available but the audio levels cannot be guaranteed.

MKP 4

SSA 424

TIF 951

UIO 256



LCP 102

HR-1

HR-2

PH-4

PH-8

PH-8000

PH3500

PH-1

PH-2

V-210

V-220

HEADSETS

HR-1 / HR-2
The HR-1 and HR-2 are medium-weight noise-reduction headsets with dynamic noise cancelling microphones, a noise reduction rating of 21 dB-suitable for use in moderately noisy environments. These models feature a unique headband design that distributes ear cushion pressure evenly over the entire ear with no points, and fold into an extremly compact shape for storage.

PH-4
Super lightweight, dual-sided for day-long comfort.High-quality monaural dynamic earphones with dynamic noise cancelling microphone on adjustable boom.

PH-8
Super lightweight, single-sided for the ultimate in day-long comfort.High-quality monaural dynamic earphones with dynamic noise cancelling microphone on adjustable boom that can be bent to practically any position.

PH-8000
Super lightweight, single-sided headset with waterproof microphone and sealed headband that make it a great choice for high-moisture environments. The PH-8000 is ideal for maximum comfort in a low noise environment, or for wearing under a helmet or under hearing protection gear. Only available with electret type microphone element.

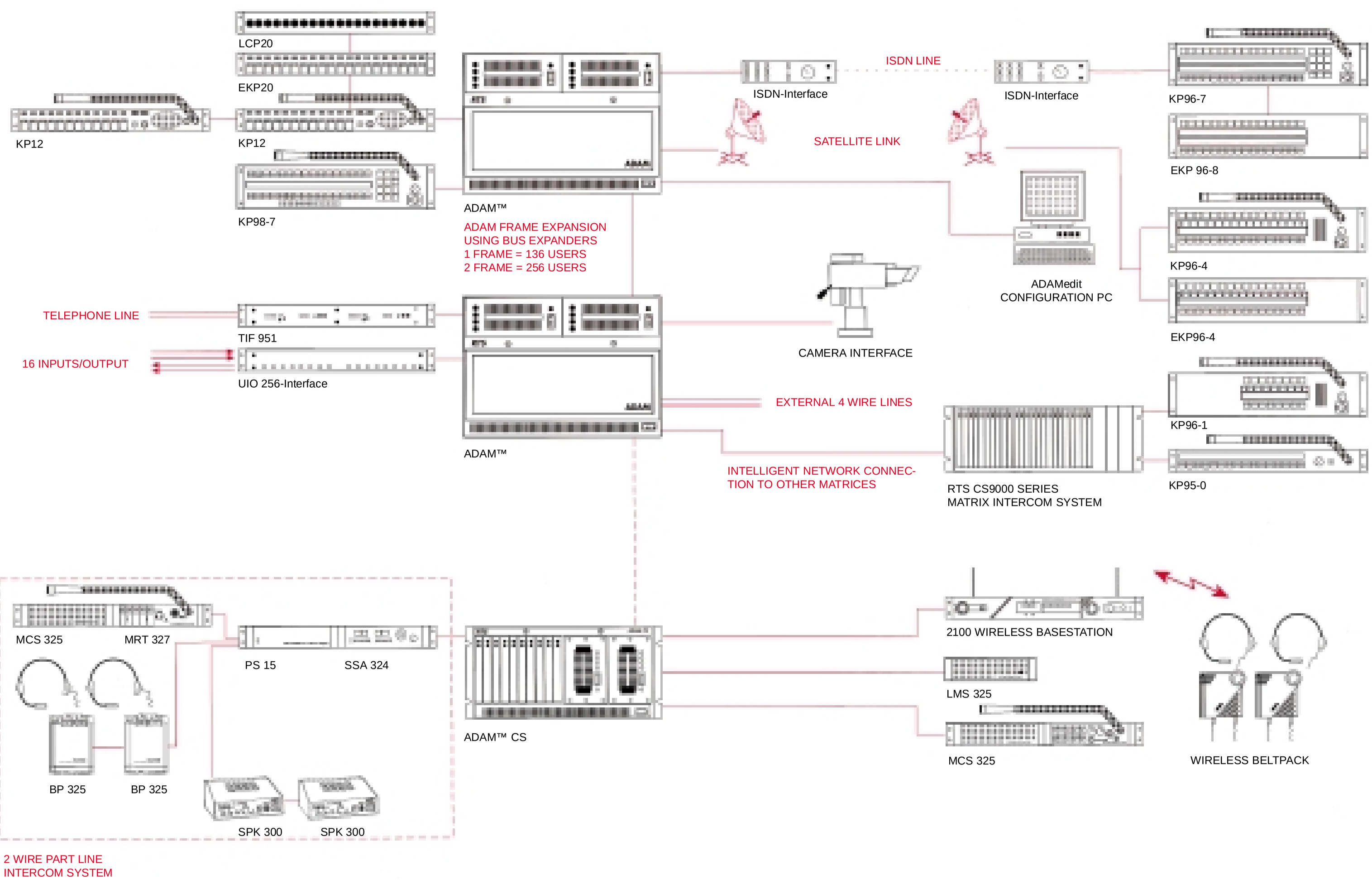
PH-3500
This powerful 13.6 oz. headset sets the standard for passive lightweight noise reduction, with an NRR of 24 db. Designed for comfort, the headset includes an easy to adjust tension system (patent pending) for proper fit and protection, as well as new headband design, stereo/mono switch and dual volume controls. Includes foam and gel ear cushions, and carrying case. Only available with electret type microphone element.

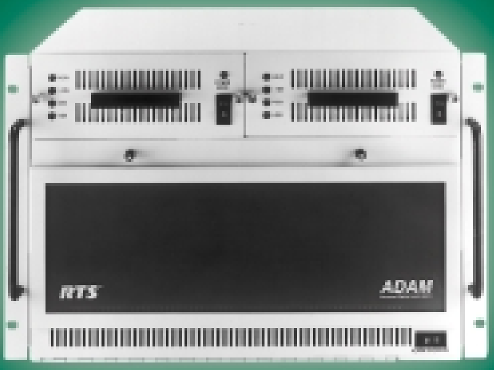
PH-1
Single-sided headset with foam-filled cushion effectively reduces ambient noise. Dynamic, monaural earphones with dynamic noise cancelling microphone attached to a continuously adjustable ball joint.

PH-2
Dual-sided headset with foam-filled cushion effectively reduces ambient noise. Dynamic, monaural earphones with dynamic noise cancelling microphone attached to a continuously adjustable ball joint.

V-210/V-220
The V-Series headsets are high-fidelity, studio quality products that may be tailored to most any need using modular cord sets and microphones. The V-210 is a single-sided headset, the V-220 is dual-sided, and both have a unique ball and socket boom with a flex-section boom for extra-flexible mic movement. The V-Series´unique floating earcups adjust to virtually any head size or shape for maximum comfort.

APPLICATIONS





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