A Comprehensive Guide to Clear-Com Analog and Digital Partyline Systems

Introduction & Acknowledgments

Clear-Com's original development of Partyline intercom effectively created the platform to coordinate live performance technical teams as now we know it. As part of our 50th anniversary, it is an appropriate time to document our Partyline communications platform—both analog and digital formats in this Reference Guide. I use the word "platform" in a very intentional way, as this Guide explains the technology in general terms and comprehensively covers the Partyline concept, as opposed to how it is handed in the specific manuals of a particular product.

Many sources were mined for creating this Reference Guide, including the Clear-Com Analog Partyline Intercom Installation Manual for written in 1988 and revised in 1997, scores of product manuals and technical training materials amassed over decades, and the Clear-Com 'Solution Finder' which resides on our website. New to this Guide is the inclusion of the HelixNet Digital Network Partyline, with the purpose of offering the same kind of broad understanding of how the two platforms may fit together for providing a single working solution. A critical aspect of this work includes documenting contemporary wiring standards and their impact on the deployment of our solutions.

Given Clear-Com's broad history in Partyline intercom, many people have contributed to this body of work, but two people must be recognized by name. First and foremost: **Charles 'Charlie' Butten** – creator of analog Partyline 'Clear-Com' technology and a continual inspiration for all intercom specialists. Charlie's contributions to Clear-Com can't be overstated. A close reading of this Guide reveals exceptionally thoughtful and effective design choices made over the years by Charlie to meet the market's changing needs. Secondly, we tip our hats to **Vinnie Macri** – our resident expert in audio and intercom applications. Vinnie took on the challenge of compiling disparate information and converting it into the key professional intercom resource you are now holding. Vinnie has been a passionate champion of Partyline technology for years. I extend my deepest appreciation to both for their efforts.

Finally, allow me to dedicate this to the entire Clear-Com community: staff and alumni, Partners, consultants, users, and supporters. Today Clear-Com has become the de facto intercom provider across many markets, all thanks to our combined passion for communications and our striving to help solve simple to complex communications needs. I consider it an honor to lead a company with such deeply-engaged and committed folks comprising our 'Clear-Com family.' It is only fitting that this be dedicated to them.

Bob Boster
President – Clear-Com, an HME Company
February 2018

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1 CLEAR-COM ANALOG PARTYLINE SYSTEMS

1.1 Introduction to Analog Partyline Intercoms

Technically speaking, an analog partyline intercom, also commonly referred to as a 2-wire system, is a communications system where the path is the same for both talk and listen. The name “partyline” (PL) came from the original telephone systems where more than one subscriber shared the same line and could hear and join all conversations at once. Therefore, partyline intercoms are always full-duplex and are commonly non-private. As a group communications tool, a partyline allows a group of people with similar workflow roles to intercommunicate all the time. An example in broadcast would be camera operators on a camera PL, or in live performance it may be stage hands. Most users only converse on one or two channels, receiving cues and talking within their own group.

“Clear-Com” as a company name was synonymous with its analog partyline system for most of the 50 years of its history. Clear-Com is a closed-circuit intercom system that consistently provides high-clarity communication in both high-noise and low-noise environments. A basic system consists of a single- or multi-channel Main Station (e.g. MS-702), or Power Supply (e.g. PS-702) connected to various single- or multi-channel Remote Stations, such as beltpacks (e.g. RS-701/RS-702) and loudspeaker stations (e.g. KB-702).

Clear-Com is a distributed amplifier system; each Main or Remote Station houses its own microphone preamplifier, headphone or speaker power amplifier, and signaling circuitry. Stations bridge the intercom line at a very high impedance (more than 10kΩ) and place a minimum load on the line. The audio level always remains constant, and does not fluctuate as stations leave and join the system. Low-impedance mic input lines (200Ω) and specially designed circuitry make Clear-Com channels highly resistant to RFI and dimmer noise.

Clear-Com stations are interconnected with two-conductor, shielded microphone cable (or individually shielded multi-pair cable as required). Portable stations are interconnected with 2 conductor shielded cables with 3-pin XLR connectors. One wire, connected to pin 2, carries the DC power (30 Volts, i.e.±15V) from a Main Station or Power Supply to all Remote Stations. The other wire, connected to pin 3, carries the 2-way (duplex) audio information. The shield, connected to pin 1, acts as a common ground. See Figure 1.

It is worth noting that every intercom circuit starts out as a 4-wire circuit; i.e., headset earphone/microphone, or separate microphone/loudspeaker.

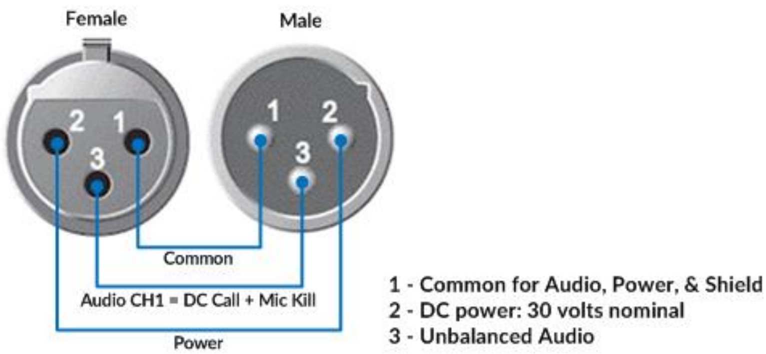


Figure 1: The wiring configurations of Clear-Com wired PL systems

Clear-Com Main Stations, Power Supplies and certain Remote Stations each have an auxiliary program input with its own volume control, which allows an external, listen-only audio source to be fed to the intercom system or local user.

Call lights are very useful to augment communications in high noise environments, backstage/green room signaling, in theatrical under stage areas, and for various industrial applications and most often to attract the attention of operators who have removed their headsets. Visual Signal Circuitry (Call Lights) is a standard feature on all analog Main stations and user Stations. Clear-Com uses a DC voltage to activate call lights.

Certain stations also have an audible Tone Alert feature which can be useful for this purpose. Outboard signal-only devices, (flashers with beeper) are also available that provide visual and/or audible call signal indication whenever a call signal is present on the intercom line. These devices are wired directly in-line to the party-line intercom channel.

Almost all user stations have Sidetone Control. Sidetone is the level of your own voice you hear while talking on the intercom. Setting a comfortable level of sidetone will ensure that the intercom sounds alive and helps you modulate your voice relative to other voices on the line. Typically, different sidetone null settings are needed depending upon whether you are using the loudspeaker. Sidetone is used for both confidence (that you are sending signal) and a warning (that you are ON).

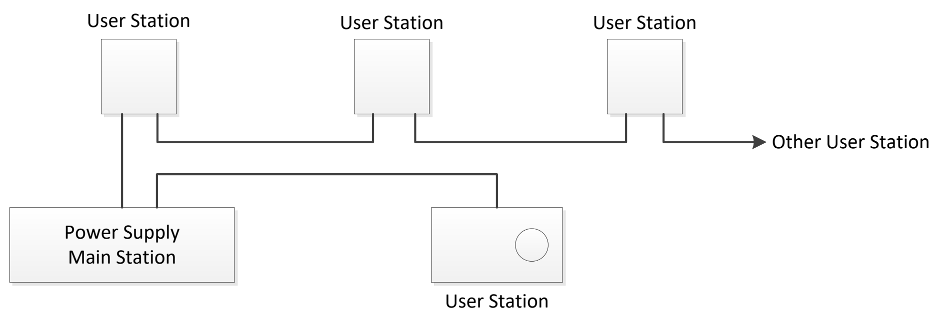
Clear-Com manufactures a wide variety of both portable and fixed-installation units and are designed to interface with other communication systems and devices.

1.2 System Components and Their Function

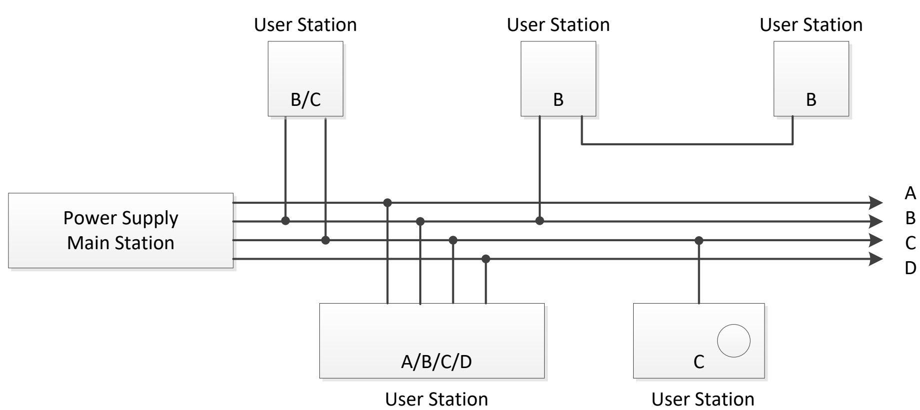
Most analog partyline intercom 'systems' consist of power supplies (or Main Stations), user stations (e.g. beltpacks, speaker stations, remote stations, etc.), interconnecting cable, headsets, panel microphones or push-to-talk microphones.

The power supply (which is normally centralized) generates the DC power for the entire system (except for self-powered user stations). The Clear-Com power supply includes 200 Ohm(Ω) system termination for the audio channel. The user station connects to the power supply and intercom line. A person connects to the user station via a headset or loudspeaker and microphone or some combination. For a given channel or channels the user stations are connected to each other in parallel. The interconnecting cable for most intercoms is standard shielded microphone cable with three pin XLR type connectors. The female XLR connects towards the power supply and the male XLR plugs into the user station. This connector scheme was chosen to prevent putting DC power onto audio microphones which also use this type cable. One exception to the use of microphone cable is where a twisted pair is the only connection between two points.

Wired analog partyline intercoms are mostly of the distributed amplifier kind. The distributed amplifier is built into the various user stations. The distributed amplifier concept allows each user to adjust their own listening level. The user station also includes a microphone amplifier, a line amplifier/buffer, volume control(s), talk switch(es).



Simple Clear-Com Loop-Thru Concept using Single Twisted Pair Shielded Cable



4 Channel Clear-Com Concept using Single Twisted Pair Shielded Cable

Figure 2: Clear-Com System Conceptuals

1.3 Power Distribution and Short Circuit Protection

A Main Station or Power Supply is the heart of an intercom system. It has special features which are not found in traditional designs. It must supply low-noise 30 Volts DC to multiple intercom lines. It must continue to operate in adverse conditions such as low AC line voltage, momentary shorts on the DC power lines to the stations, and excessive peak loads during power-on conditions.

The power supply or Main Station generates the DC power (typically 28–30 VDC), for the entire system. In analog partyline systems the DC voltage is applied to the XLR3 pin-2 audio conductor.

The question of how many Speaker Stations and Beltpacks can be powered in varying combinations by a Main Station or Power Supply is dependent on many factors, most significantly the type of cable, its construction and electrical characteristics and how much of it is being used. Refer to Section 5 Digital & Analog Partyline Intercom Cabling Comparison for a more detailed explanation.

Additional power supplies can be connected for additional power capability if required by the application. The current draw requirements of Clear-Com Remote Stations and Beltpacks vary with model and use. A station which is simply "on" and not being used may draw only a small amount of current.

The total current capability of a Main Station or Power Supply is split between its channels. Short circuits and overloads on either channel will not damage a Main Station or Power Supply. It will simply cut power off to each channel which exceeds its maximum current or which causes the system maximum to be exceeded.

Clear-Com's fail-safe design automatically shuts down the power to a channel when a short circuit or electronic overload is sensed on that channel. The other channel will continue to operate normally. Once the fault condition is removed, the fail-safe circuit will restore power, even under full load conditions. LED indicators signal a short or overload on either channel.

The DC power output details for Power Supplies/Main Stations are:
- 1.2A (Amperes) continuous output
- 2A (Amperes) peak output (not exceeding the 1.2A rating for more than 2 seconds per 1-minute period)

The station internal power supply senses the difference between short-term and long-term shorts and overload conditions. Figure 3 shows that after the first few times a short or overload occurs, the power supply will try to restore power after only 0.5 seconds. If the short or overload persists or occurs repeatedly, the power supply will take progressively longer (to a maximum of 20 seconds) to try to restore power. This protects the power supply from damage due to overheating. Once the short is removed, the channel will recover, even under a full load condition. The automatic power restore times are shown in the following chart (Figure 3):

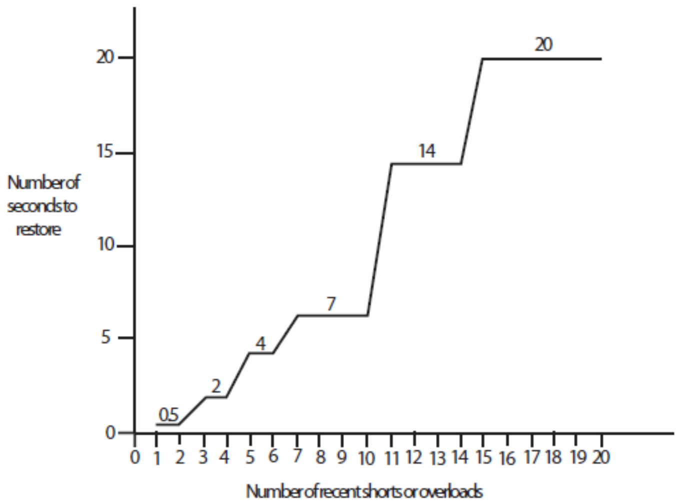


Figure 3: Automatic Power Restore Times

A channel which is within its maximum current rating will not be affected by another channel being automatically cut. Within 15 seconds of automatically cutting power to an overloaded channel, the Main Station or Power Supply will attempt to turn power on again. This allows momentary short or overload conditions to clear automatically. Red Short LEDs corresponding to each channel light if power is cut, indicating which channel is affected. This indication will assist in locating the shorted or overloaded channel. Shorts are generally caused by faulty wiring or damaged cables. Overloads are sometimes caused by connecting too many beltpacks and stations to a channel.

To ensure proper power is budgeted for each Partyline unit, you can refer to the Power Supply Calculator that is available on the Clear-Com website at:

https://www.google.com/url?q=http://www.clearcom.com/download/technical-documents/encore-power-supply-calculator&sa=U&ved=0ahUKEwjala2u5vLWAhULxFQKHUZEA20QFggEMAA&client=internal-uds-cse&usg=AOvVaw0qnZLWzeZ8woaNjUjzwyGH

The following (Figure 4) is an example of the Power Supply Calculator in use:

1. Enter quantity of each power supply
2. Enter quantity of each accessory
3. Green indicates how many AMPs under budget
4. Red indicates how many Amps over budget
5. If Red, reduce accessory quantity to be under budget
   or Increase Power Supply quantity to be over budget
6. If Green, you are within the power budget

Power Supplies	Max Current (AMP)	QTY	Power Budget
PK-7	0.400	1	0.36
PS-702*	1.800	1	1.62
PS-704*	1.800	1	1.62

*sum of all channels   PS Total Budget *3.6*

AMPs Over/Under budget: *2.225*

Accessories   Acc Total Budget *1.375*

Accessories	Max Current (AMP)	QTY	Power Budget
RS-701	0.028	1	0.028
RS-702	0.040	1	0.040
RS-703	0.040	1	0.040
RM-702	0.110	1	0.110
RM-704	0.195	1	0.195
KB-701	0.110	1	0.110
KB-702	0.110	1	0.110
KB-702GM	0.110	1	0.110
HB-702	0.100	1	0.100
HB-704	0.100	1	0.100
TW-12C	0.050	1	0.050
IF-4W4	0.050	1	0.050
EF-701M	0.080	1	0.080
TW-47	0.060	1	0.060
TWC-701	0.030	1	0.030
RS-601	0.050	1	0.050
RS-602	0.062	1	0.062
RS-501	0.050	1	0.050

Figure 4: Power Supply Calculator available on Clear-Com website

1.3.1 Electrical Isolation for Partyline Intercom Systems

The MT-701 partyline isolation box is an accessory for Clear-Com analog partyline systems that will electrically isolate the ground, power, and signaling of two intercom systems. If two power supplies or main stations are connected together in a system but located in different buildings or on different AC power systems, an objectionable "hum" may be caused by a ground loop. The MT-701 will eliminate the "hum" by breaking the ground loop between the powered stations.

This device also allows channels from two separate Clear-Com systems to be connected together and because of the power isolation it provides, each system can be shut down independently. In addition, the system isolator will pass the “Call” signaling from one system to the other with complete electrical isolation. The MT-701 offers transformer isolation for the audio signal and opto-isolation for the “call” signal. It also provides the termination for both systems.

Tip: *The MT-701 must have power on both ends in order to carry a call signal. However, power is not needed on either end for an audio-only connection to work. Only one termination is required between the connected systems.*

System Termination - Isolation Box relationship
One system isolation box must be used for each channel of the intercom systems being connected together. The system isolator box becomes the termination for the connected channels from each system. The station- or power-supply- based terminations for these channels must be turned off. Power is provided by the attached intercom stations or power supplies. The MT-701 is intended for standard Clear-Com intercom lines only and cannot be used on “TW” (Telex RTS) powered lines.

1.3.2 Termination of the Intercom Line

The fundamental concept of Clear-Com Partyline intercom is that all channels are terminated in one location, preferably at a Main Station or Power Supply. The Clear-Com system termination is not electronic but a passive resistor. This audio termination circuit is what allows many intercom user stations to be connected to a single partyline. Each channel of the system would have its own terminator. The termination prevents drastic changes in the impedance of the PL channel if remote stations or beltpacks were added or removed from the PL line. One Line Termination per channel is needed throughout the intercom network, and is usually located in the Main Station or Power Supply.

Note: All intercom lines must be terminated. Care must be taken not to "double-terminate" a line. All unused intercom lines must also be terminated.

Switching of the channel terminations ON and OFF is done with switches or jumpers on the Main Station. In most systems, the terminations should be in the ON position (default setting). See Figure 5

Clear-Com Power Supplies also provide switch-selectable termination networks on all intercom lines except the small PK-7. The PK-7 has an internal 200ohm termination permanently set to on. The termination cannot be turned off, so if it is to be used along with a main station or power supply the termination for the channel the PK-7 is connected to should be set to the "off" position as the PK-7 is now providing the termination for that channel.

It is up to the user to ensure that the terminations are set correctly. An unterminated line will cause excessive levels, possible oscillation of line drivers, and squealing in the headsets. An intercom line with double or multiple terminations will cause low levels and the inability to null the circuit. The termination switches on a Main Station should be set to the OFF position only if the channel is terminated by another Main Station or Power Supply in the system.

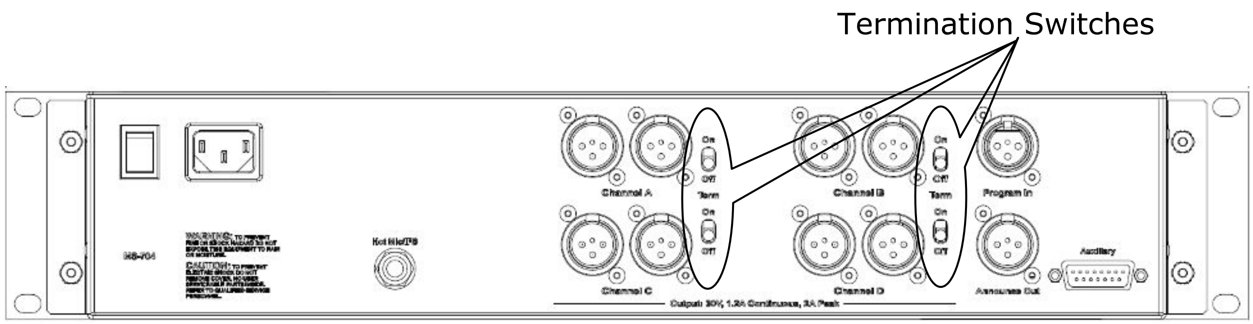


Figure 5: Rear of MS-704 Main Station showing Termination Switches

Note: If there are no other Main Stations or Power Supplies terminating the line, the termination switch on each channel of the Main Station should be switched to ON.

1.3.3 Check Termination

A. Using a multimeter, measure the resistance between pins 1 and 3 on one of the Channel A XLR connectors at the rear of the unit.

B. If the channel is terminated properly, then the resistance should measure approximately 4,000 – 5,000 Ohms. A very high channel resistance (over 10,000 Ohms) means the channel is not terminated. Channel resistance of 2,000 Ohms indicates a double-termination. If a double-termination is indicated, locate the other terminated Power Supply or Main Station and set its termination to OFF.

C. Repeat for the other channels.

D. Check resistance between Chassis Ground and pin 1. Using an Ohmmeter, measure the resistance from pin 1 on the Main Station or Power Supply to chassis ground. The measurement should read 10 Ohms. A high reading (over 100 Ohms) indicates that a component failure within the system and result in an audible "buzz" in the system. A reading of less than 10 Ohms (or a short) typically indicates that the shell and pin 1 of one of the interconnect cables are shorted together.

1.4 Stations

The following Figure 6 is a listing of every Clear-Com device produced since the beginning itemized by model. This includes not only the intercom devices but also the interfaces and interconnect devices used to connect to other communication systems. All products are referenced in following subchapters.

Clear-Com Analog Products by Category and Model

Older Models → Newer Models

MAIN STATIONS	Portable	CS-100	CS-200	CS-210	CS-222	CS-702				
MAIN STATIONS	2-Channel	CS-100K	MS-200	CS-200K	MS-222	MS-232	MS-702			
MAIN STATIONS	4-Channel	MS-440	MS-400A	MS-704						
MAIN STATIONS	4-Channel Swithboard	SB-412A	SB-440	SB-704						
MAIN STATIONS	8-Channel	MS-808	IS-808							
MAIN STATIONS	12-Channel	MS-812 with IP-1200 Interconnect Panel								
MAIN STATIONS	Remote	RM-3000	RM-220	RM-440	RM-400A					
BELTPACKS	Single Channel	CP-100	RS-100	RS-100A	RS-501	RS-601	RS-701			
BELTPACKS	2 Channel	CP-300	RS-201	RS-502	RS-522	RS-602	RS-622	RS-702		
BELTPACKS	2 Channel TW (includes RTS)	RS-502-TW	RS-522-TW	RS-603	RS-623	RS-603R	RS-623R	RS-703		
2-WIRE IFB	Systems Electronics	PIC-4000B	PIC-4704	PIC-4744						
2-WIRE IFB	Systems Controller	MA-4, AX-4	MA-704	AX-704						
2-WIRE IFB	Receiver	TR-50T	TR-532 Stereo/Split-Feed Receiver							
OTHER	Power Supplies	PS-300	PK-5	PS-100	PS-20	PS-452	PK-7	PS-702	PS-704	
OTHER	Remote Speaker Station	KB-100	KB-111A	KB-112	KB-211	KB-212				
OTHER	Headset Station	MR-102A	MR-202	MR-204	HB-702	HB-704				
OTHER	Hybrids/2W-to-4W Interface	AC-10K/H	EF-1M	AC-701	IF4-B	TW-12B	TW-12C	EF-701M	IF-4W4	TWC-704
OTHER	Cameras	ICP-4 ISO Control Panel		ISO-4000 ISO Central Electronics						
OTHER	Commentary/Announce Box	AB-100	AB-120							

Figure 6: Clear-Com Product listing by category and model
(oldest model is listed first)

1.4.1 Main Stations

The Main Station is designed for applications in which a person needs to communicate with other people or groups of people, either individually or simultaneously, while still maintaining isolation between each of the channels. This station is usually given to a director, producer, stage manager or show caller depending on the workflow.

The Main Station is a multi-channel microprocessor controlled station intended to work with other Clear-Com analog partyline products. It is a self-powered unit and can also power remote Main Stations, speaker stations or beltpack headset stations. It is normally used as a Main Station, (which provides system termination at a central location), but may be used also as a remote station that provides extra power capacity.

A Main Station will have the following features: Individual talk, listen, and call pushbuttons for each channel. Each channel has individual listen level controls, channel null adjustments, and multiple program inputs with program feed level controls. Main Stations have the ability for Interrupt Fold Back (IFB) of program feeds to intercom channels on talk or call and an all-talk button access to all channels.

Main Stations have extra features for special tasks such as main program feed to the stage announce output with interrupt relay closures, front-panel remote mic-kill switch to turn off all latched talks, a rear panel jack providing a buffered un-switched output of the selected microphone to be used with an external IFB system.

Present models include the CS-702, MS702, MS-704 and SB-704. Discontinued models are: CS-100, CS-200, CS-210, CS222, CS-100K, MS-200, MS-222, MS-232, MS-400A, MS-440, MS-808, MS-812, SB412 and SB-440

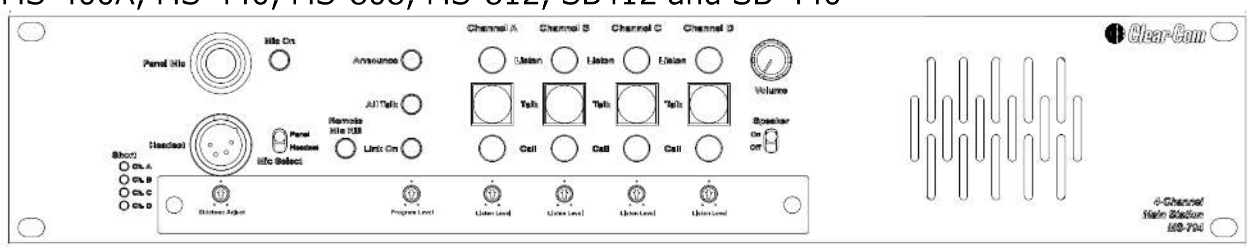


Figure 7: MS-704 Main Station Front Panel

1.4.2 Beltpack Headset User Station Functional Description

A typical single channel beltpack headset user station has the following connectors: Intercom line and loop-thru (XLR-3MF) and a Headset Connector (XLR-4). The 3-pin XLR-type female and male intercom connectors connect to the Clear-Com intercom line. They are wired in parallel. While the female connector is normally used as the input, either one may be used as the input or as a loop-through output.

Figure 8: RS-700 Series Beltpacks
(Single Channel/Two-Channel/Two-Channel TW)

The station has the following controls: Microphone ON/OFF (called a TALK switch). The talk button turns the microphone circuit on and off. It also determines whether the circuit is switched on momentarily or continuously. For momentary action, press the talk button for the desired length of time needed to talk on the connected headset’s microphone, then release to deactivate. For continuous (or “latching”) action, tap the talk button twice (this depends on the era of product in question -

newer RS-700 Series use a single tap) in quick succession to lock the microphone on. To turn the microphone off, press and release the talk button again. The RS-500/RS-600 and RS-700 models have Latch disable modes.

It may also have a call lamp and a call lamp send button. When the call button is pressed, it transmits a call signal on the intercom line that illuminates all other stations' call indicator lights on that channel. Units also have headset Volume Controls. The volume control adjusts the listen level of the incoming intercom signal. The single channel requires a single volume control. The two-channel units have a volume control for each channel. The incoming audio level has no effect on the outgoing talk signal. Its range is from full "off" (counterclockwise) to full "on" (clockwise). Units built in 2001 and later contain an output limiter. Examples of earlier stations are the Clear-Com RS-100A, RS-501, RS-601 and RS-701.

A typical two-channel headset beltpack user station adds a channel selector switch to the above, and has the following connectors: Intercom line(s) and loop-thru (XLR-6MF) and a Headset Connector (XLR-4).

The 6-pin female connector connects to the Clear-Com intercom lines. The RS-622 does not have loop-thru to avoid confusion with the 6-pin male headset jack. Examples of this type of station are: Clear-Com RS-201, RS-502/RS-522, RS-602/622 and RS-702. Newer units have two talk buttons, two volume controls, two call buttons, and two status indicators to tell which talk button is engaged.

Compatibility with Two-Wire (TW) Intercom Systems

Clear-Com designed the RS-502/522 TW, RS-603, RS-623, RS-603R, RS-623R and RS-703 series of beltpacks to operate with "TW" (two-wire) systems. These beltpacks have the appropriate audio levels and frequency response curves for operating on a TW intercom line, as well as having such TW features as AC call signaling and remote microphone shutoff. The RS-603, RS-623 and RS-703 models are shipped ready to operate on *Clear-Com/TW* intercom lines, while the RS-603R and RS-623R models are shipped ready to operate on competitor RTS-TW intercom lines.

However, using a beltpack's onboard controls, you can easily reprogram any of these models to work on the opposite system. For example, you can program a Clear-Com TW-compatible beltpack to operate on an RTS intercom line, and program an RTS-compatible beltpack to operate on a Clear-Com TW intercom line.

> ***Note: If you install an RS-603 or RS703 series beltpack with standard Clear-Com equipment, channel B functions, but channel A does not. The use of the TW-701 interface is required.***
>
> ***Note: All TW compatible units use the 3pin XLR loop-thru connector except the RS-502/522 TW which are equipped with the 6-pin female connector. (The RS-522 does not have loop-thru to avoid confusion with the 6-pin male headset jack.)***

Note: The RS-603 and 703 support both DC and AC call signaling in order to be compatible with RTS systems. Because the standard Clear-Com intercom "call" signal function is normally accomplished by applying a DC voltage to the intercom audio line, call signaling is deleted from the Channel A line of the RS-603/703 unit.

This means that a RS-603/703 beltpack connected to a MS-704 unit via a TWC-701 unit will only be able to send and receive call signal on channel-B. The same is true for the RM-704 unit it will only be able to send and receive call signal on its channel B. However, both units will be able to talk/listen on both intercom channel A and B.

Note: Only TW enabled products should be connected to TW PL systems otherwise non-TW beltpacks, (500 Series), will load the DC power on pin 2 reducing volume level and changing sidetone characteristics on that channel. The exception is that the RS-601/ RS-701 beltpack has been specifically designed not to load channel A of TW systems while still being able to talk and listen to channel B of the TW system.

1.4.3 Speaker & Remote User Station Functional Description

Speaker stations are often installed with an electrical back-box so they may be used as in-wall user stations and can also be found surface mounted. Clear-Com offers an enclosure for desktop or surface wall mounting. This combination of speaker station with box is most recognized as a 'biscuit'.

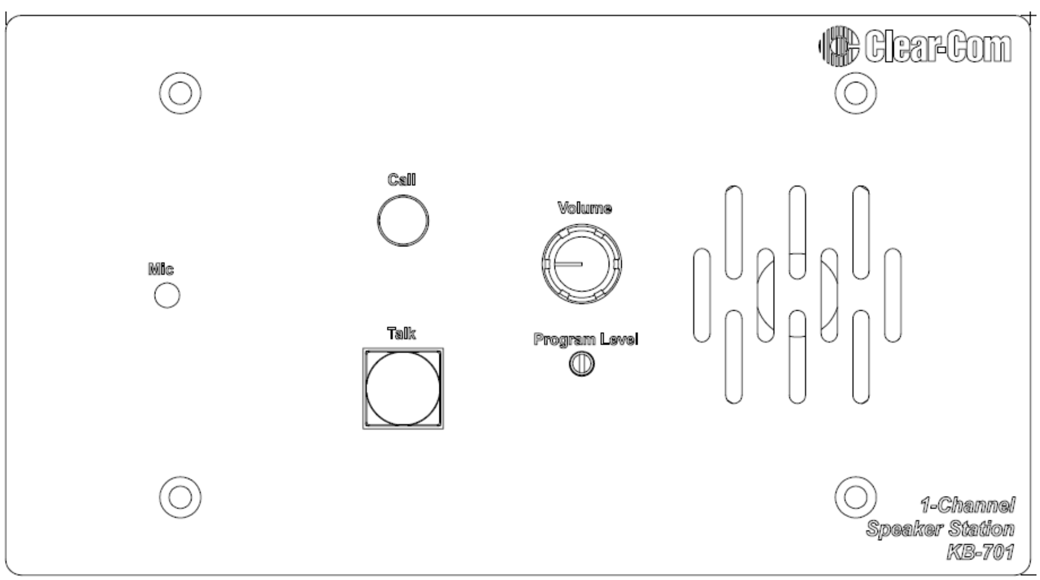


Figure 9: KB-701 Speaker Station example

Speaker stations provide intercom communication capability in places where wearing a headset may not be feasible (although some speaker stations are supplied with a headset connector). Remote speaker stations permit the user to select between one or two channels of party line communications, with the ability to talk and/or listen on the selected channel. The user can listen via the integrated speaker, or may use a headset or telephone-style handset—and can talk via a headset mic, gooseneck microphone, a telephone handset, or a push-to-talk microphone. Speaker stations offer both visual and audible call signaling to attract the attention of operators. The Remote Mic Kill (RMK) feature on Main Stations will turn off any open microphones on speaker stations.

With single channel speaker stations, such as the KB-100, KB-111, KB-112, KB-211, KB-212 and KB-701, local or remotely controlled talking or listening is selectable and allows either hands-on or hands-free operation.

In some installations, the 2 channel KB is intended to be used only with one intercom channel. Setting the single channel switch to the on position will connect the station to the one intercom channel regardless of the position of the front panel channel selector

The KB single channel devices feature "control logic" CMOS circuitry for programming the operation of the station. This allows remote or local control (or both) of the speaker and the mic. The station operator pre-sets a bank of dip switches that are located on the electronics module. If pre-set for remote control, the unit speaker and/or mic can be activated by all other Clear-Com stations (on the same channel, using the visual signal circuitry).

The single channel KB operating modes are:

- NORMAL: Speaker is on. Mic is locally activated by pushbutton on front panel.
- REMOTE PAGE: Speaker is off except when turned on by remote control, or used to page anyone at that KB’s location. Mic is activated locally.
- REMOTE LISTEN: Speaker is on. Mic is turned on locally OR by remote control, which allows that KB operator to talk "hands-free."
- REMOTE LISTEN-PAGE: Speaker is normally off. Mic remains on for hands-free talking. Another station can turn off the mic and turn on the speaker for paging that KB operator

The *Talk* button operates in momentary mode. The integral microphone and speaker offer half-duplex communication.

All KB single & dual channel units feature a program input which allows the monitoring of external audio through the speaker or headset. A program level trim control is provided to set the program audio level heard in the panel speaker or headset. This program input can also be used as a paging function.

All KB single & dual channel units except KB-100, KB-111, and KB-112 have audible tone alert. This can be useful when the operator’s attention has been drawn away from the KB indicator panel. This can be enabled to sound when a call signal is received on the selected channel or either channel. The audible tone alert level can be adjusted or turned off by an internal control. The tone alert will not sound if a call signal originates at the KB station or if the speaker on/off switch is turned off.

A 4-Wire option module is available for speaker stations. This option allows the speaker or headset station to be operated over longer distances than are possible with a standard Clear-Com partyline intercom line. This option can also be used to connect with fiber optic transmission equipment. The 4-Wire Option module connection is a transformer-isolated, audio-only connection and does not convey the call signal. The module also requires local powering. Two channel stations can have two separate 4-wire connections with this option. The Channel Switch on the front panel selects which channel is connected.

1.4.4 About Headsets and Microphones for Analog PL

Intercom stations, whether they are analog or digital partyline or more sophisticated matrix user key-panels, have been generally designed for use with dynamic headsets. A dynamic headset typically incorporates a noise cancelling dynamic microphone with an impedance range of about 150 to 500 ohms depending on the manufacturer. An electret microphone headset is usable with an analog partyline intercom and may be self-sensing (i.e., internal circuitry senses an electret element) or may have to be selected for use.

Intercom station panel microphones are typically electret design and have an impedance of 300 to 2000 ohms, and are designed to be phantom-powered with a voltage range of 1.5 to 15 VDC. These panel microphones are typically gooseneck, and are offered in various lengths and with differing connectors (e.g., ¼ inch TRS and/or DIN screw in).

Push-to-talk handsets and telephone-type handsets are useable in any partyline intercom system but more practical to users who cannot use a headset because of their workflow. These units tend to have limited frequency response of 300–3300 Hz. The limited high frequency, although being sufficient for voice grade communications, may not be suitable for users who maintain long periods of listening because it creates fatigue.

Earpiece impedances in headsets range from 50 ohms to 1000 ohms. The use of a 50ohm headphone requires a headphone amplifier capable of powering the line. Lower impedance loudspeakers will draw more current from the amplifier. This becomes more important when we talk about IFB earpieces that may have extended audio bandwidth that includes low frequency content that excites the loudspeaker to pull more current. While low impedance headsets are a good choice to provide enough SPL (sound pressure level) to overcome the interference from loud environments, caution should be used to avoid distortion and most important, ear damage. One option for high noise environments are headphones that offer an acoustic isolation of 20dB or more to protect the user. Headphones from manufacturers are typically labeled lightweight or full cushion, and offered as single-sided for one earphone or dual-sided with two headphones. It is important to note many user stations including matrix key-panels and wireless user stations have a headphone impedance range from 25 to 600 ohms.

Binaural/Split Ear Function

To be considered, Binaural/Split Ear requires a stereo headphone or 'dual muff' headset. The intercom listen function is by and large set up as mono, i.e. a user listens to all intercom channels, and any program if inserted, in both left and right earphones. This is also called dual listen. The dual muff headset will be provided with a 4-pin XLR wired as monaural headphones as shown in Figure 10.

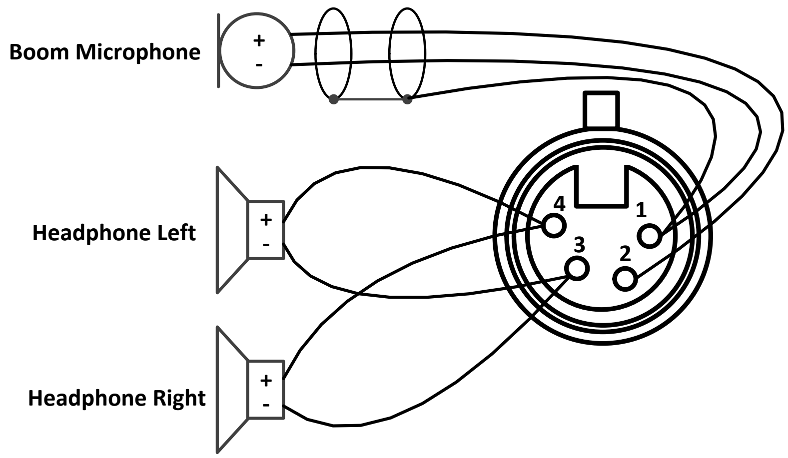


Figure 10: Dual Muff Headset to 4pin XLR - Mono

The purpose to Binaural/Split Ear operation places an intercom channel in one ear and another intercom channel in the other ear– the operation resulting in listening to 2 sources separately and independently. In this case, the dual muff headset will be provided with a 5-pin or 6-pin XLR wired as stereo or split headphones as shown in Figure 11.

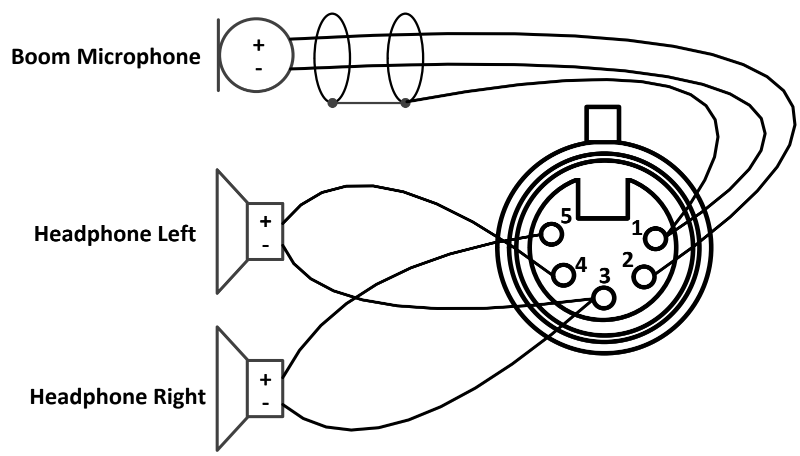


Figure 11: Dual Muff Headset to 5pin XLR – L/R stereo

The RS-622/623 beltpacks were the first units whereby a user could separate the two channels

1.5 Source Assignment

Following from the initial design of analog partyline by Clear-Com, many products follow the model of 'one channel per cable'. So, if a cable is being used as channel A on it and you need to be on channel B, you must find that cable and re-patch—unless you have a multi-channel user station or source assignment panel.

Source assignment panels are manually operated matrix switches. A source assign panel is usually a separate piece of hardware, although Clear-Com offers a slider matrix source assign feature in the SB704 4-channel Main Station. Applications include any facility with at least four channels of partyline and twelve or more "drops" or "buses" or "circuits," and where the assignment setups change on a regular basis.

Inputs to the source assign panel are sensibly called "sources" and are typically the partyline channels from a power supply or Main Station or partyline interfaces from a matrix or similar system. The "destinations" are the partyline circuits or buses themselves; i.e., cabled outputs from the source assign panel that are "wet" (have supply voltage, 24–30 VDC). Cables are typically 3pin shielded XLR microphone cables. Remote intercom stations, beltpacks, or other interfaces that can also be groups of such devices connected together are patched to various circuits/buses. The source assignment panel offers flexibility in assigning intercom channels to intercom buses and provides the ability to easily reassign channels to buses on the fly.

The MX-840A Assignment Panel (discontinued) was a mechanical device that could switch up to 40 stations to 8 channels. This was replaced with the RCS-2000 (discontinued) and then replaced with the current RCS-2700. The capacity for the RCS-2700 can be expanded to 72 destinations across 15 channels.

With the advent of digital partyline systems, the need for source assignment hardware has been eliminated as all channels now are found on one cable and channel assignment is selected inside the user stations or browser configured.

1.6 Interfacing to Partyline

Interfacing generally involves interconnection of separate production communication systems. The most common interface method to wired analog partyline systems is to convert the 2-wire signals to 4-wires. The challenges with interfacing intercoms are conversion, level matching, and signaling. The best means for conversion is the 2-Wire-to-4-Wire converter also known as a "hybrid".

Hybrids/2W-to-4W Interface

The term hybrid refers to a device that converts two-wire to four-wire audio and vice versa. Analog hybrids initially used transformers, as shown in Figure 12, and later used op-amps to convert between two-wire and four-wire audio. Digital hybrids with digital signal processing (DSP) chips are most commonly used to perform two-wire/four-wire conversion.

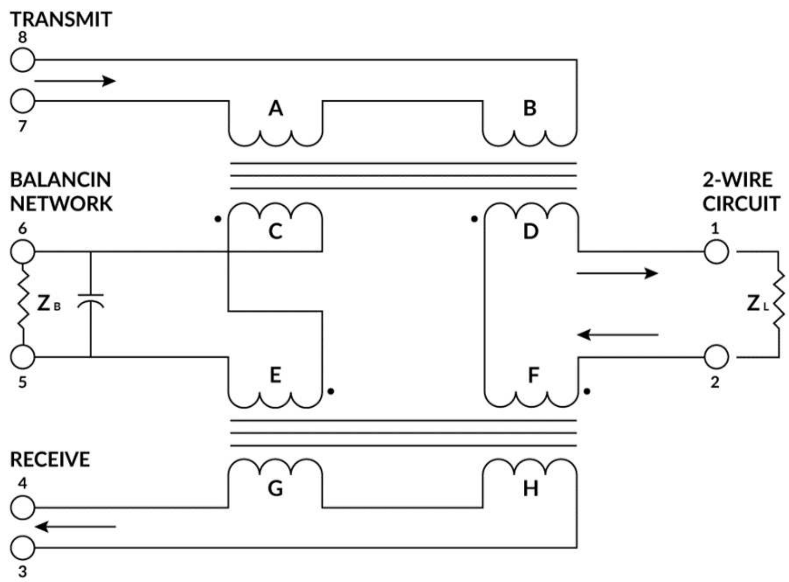


Figure 12: Transformer Analog Hybrid

In general, the performance of hybrids is an area of concern and we evaluate this with a metric called 'trans-hybrid loss'. This is a measure of the loss or isolation between the transmit and the receive ports on the four-wire side of the circuit – in this case 'loss' being desirable up to some amount. Trans-hybrid loss depends on signal cancellation accomplished through defining the line impedance and mirroring it in a balanced network. The variations in impedance presented on the two-wire side of the hybrid make balancing difficult and often result in poor trans-hybrid loss.

Hybrids are also used in intercom systems as interfaces to telephone networks and as interfaces between two-wire and four-wire intercom systems. The inherent problems associated with two-wire to four-wire conversion justifies why it is better to use the four-wire interface that is available on most camera control units (CCU) instead of a two-wire interface when connecting CCUs to a matrix intercom system. More details on camera interfacing follows.

Null: Nulling refers to adjustments made in balancing a network to achieve greater trans-hybrid loss by making resistive, capacitive, and inductive adjustments to match the impedance of the two-wire side of the circuit. Inductive, resistive, and capacitive nulling affect low-, mid-, and high-frequency bands. Modern hybrids are digital, thus capable of auto-nulling.

A hybrid's ability to isolate the transmit signal from the receive signal in the 2-wire-to-4-wire interface is critical. The quality of this isolation is technically known as return-loss. A side tone nulling control fine tunes the circuitry to best match the devices to the acoustic conditions near the intercom, as well as to the electronic conditions on the intercom line. They should be set at the time of system installation and adjusted as changes are made to the system like adding additional cables or more than a few additional end points or interfaces.

Four-wire audio is defined as a pair of conductors carrying an input/receive signal and a second pair carry the output/send signal. This hybrid circuit connects the four-wire audio to the single wire in such a way as to variably restrict the user's reception of his own voice on the intercom line, which is often referred to as 'side tone'. There should be no component of the talk signal in the listen signal. This is accomplished by adding an inverse polarity copy of the talk to the Listen. The level of the inverted talk signal must be exactly the same amplitude as the 2wire circuit. High gain between the send and receive poses a risk of oscillation or 'howling' resulting from acoustic and/or electronic coupling within a headset or between a speaker and a microphone.

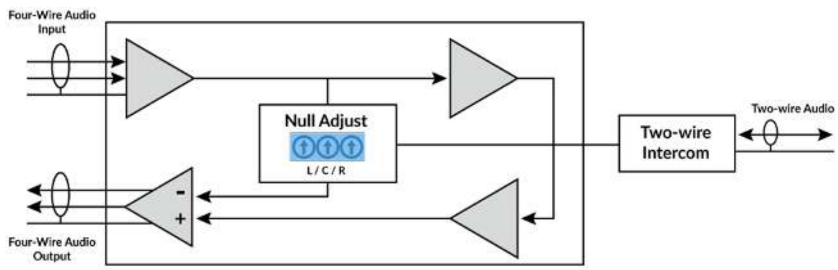


Figure 13: 2W-to-4W Interface with L/C/R Null Circuit

With manual nulling devices, there are the following accessible controls: Separate "R" (Resistance), "L" (Inductance), and "C" (Capacitance) controls compensate for each component of the line's impedance, providing the best null possible. Ideally, there should be no portion of the talk signal in the listen signal. The variation of the 2-wire line phase coherency is affected by the cable capacitance (length of wire and gauge) and inductive elements of the line. The "C" control compensates for cable capacitance; the setting depends on the length of the line. The "L" control compensates for the low-frequency inductive and capacitive elements the wiring of the external partyline presents to the line.

The importance of 2-wire termination, (as explained earlier), lack of, or double termination also influences the null result. Every time communication goes through a 2-to-4 wire conversion (hybrid) the signal is degraded somewhat.

Cameras
A common requirement for interfacing intercoms is video camera's communication circuits, often added by manufacturers in an attempt to simplify getting cues to the camera operators. However, this effort to be helpful is hampered by a lack of standardization. There is no standard design for camera intercoms among different camera manufacturers. In addition, some manufacturers don't even use the same intercom design within their own line of cameras.

The majority of large installations and virtually all mobile units use camera isolate systems. A camera isolate system allows an operator (usually the video operator or technical director) to have private communications with any of the camera operators for set-up or maintenance purposes without interfering with ongoing communication between the other cameras and the director.

Cameras and analog PL systems
The best way to interface a camera intercom to analog partyline systems is to access the Camera CCU intercom circuit as a 4-wire circuit, and then use a 2-to-4 wire interface to convert to partyline. A 4-wire connection is frequently not available on lower cost cameras.

If a 4-wire connection to the camera intercom is not available, almost always the next best choice is to interface to the headset connection on the camera CCU/DCU. (The headset connection must be a 3 or 4-wire circuit.)

It is always best to interface each camera individually. Trying to combine the 2-wire partyline circuits the Camera Control Unit (CCU) or Digital Command Unit (DCU) provides into a single partyline and then use an interface to convert the camera partyline to Clear-Com or RTS partyline is the least effective way to interface multiple cameras. There are several reasons:

- First, the more cameras you connect in parallel, the worse the partyline impedance characteristics become, therefore a good null is difficult, if not impossible, to obtain.
- Second, if the impedance characteristics of a single camera are worse than the other cameras (which is not uncommon), the overall partyline impedance characteristics are degraded to the level of the worst camera.

***NOTE: Many camera CCU/DCUs have the intercom connection on a 3-pin connector, or on three pins of a multi-pin connector (frequently along with the "tally" controls). The pin-out is frequently identified as: +/hot, -/cold & common. Many people assume that this indicates a 3-wire intercom circuit, with one pin "send", one pin "receive", and one pin "common" to both. This is not true. The +/hot & -/cold pins are a 2-wire partyline, with the "common" simply being a shield/ground. There are no known exceptions to this. These problems can be eliminated completely by bypassing the camera altogether. Run a microphone cable with the camera cable, and plug a beltpack in at the end. You see this commonly done at a remote production such as a news shoot or sporting event when there is a set on the field*.**

Two-Wire IFB Systems
Two-wire IFB systems generally require a central electronics control unit to allow multiple users to select one of many IFB channels to talk to different talents. An example of a two-wire IFB system is shown in Figure 14.

The circuitry in the central electronics unit allows for the selection of program sources and the switching of those sources to an IFB channel. When a user talks on an IFB channel, that interrupt audio is mixed with the program source audio, which is dipped in level according to a user-defined setting.

The IFB control unit also superimposes a DC voltage on the IFB line to drive the talent listen receivers. These IFB receivers can, depending on model, be either a two-channel model that can provide non-interrupt audio in one ear and program with interrupt (IFB) audio in the other ear, or a single channel model that provides only interrupt audio.

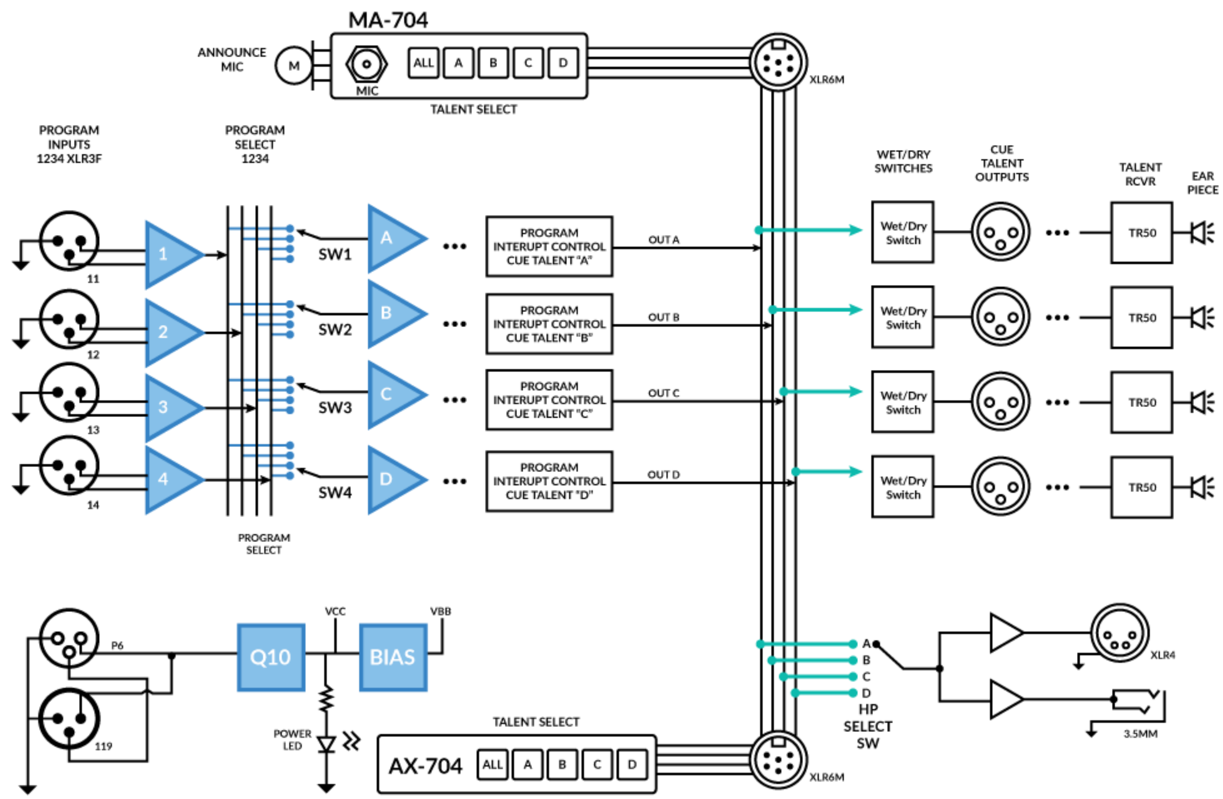


Figure 14: Example of Partyline IFB

The interrupt audio is generated at IFB control panels that are interfaced to the IFB controller or from a user station. User stations can also generate a local IFB where the program audio that is fed into the user station is interrupted by the talk at the user station and is sent down an intercom channel. A locally generated IFB does not have the flexibility of a centrally based IFB system since only that user station can talk on a given IFB channel. With central IFB systems any number of users can potentially cue talent on the same IFB channel. In a digital matrix system, IFB priorities can be set so that one station has priority over another station for talking on an IFB channel, however in partyline-based systems additional presses on the same channel simply add into the same talk path. User panels can be programmed with different priority levels to ensure that the most important producer in a production will always have communications access to talent.

The central electronics unit provides tally information so that an IFB button assigned to a channel will illuminate to indicate when that channel is in use. For example, if a user talks on IFB channel 1, the IFB 1 button on every user station will illuminate to indicate that IFB channel 1 is in use.

IFB control panels and appropriately equipped user stations have an IFB all call button that will allow a user to speak on all IFB channels simultaneously. This is a handy feature when a producer needs to provide the same cue to multiple talent on a set.

The monitor speakers in control areas can be connected to the IFB system such that they are muted or dimmed when the IFB is active. This will prevent acoustic leakage of control room monitor audio into the IFB during an interrupt.

Commentators

This term refers to a radio or television team member who commentates on the events and actions of sport or news. Other common terms are announcer, newscaster, presenter, or reporter. Within remote sports productions they are referred to as sportscasters, analysts, play-by-play and color commentators. Some sporting events require numerous inputs from specialists, statisticians and other external cueing sources.

Commentators may work from the event itself, or in some cases 'off tube' via the host signal while based in their home nation. 'Off tube' refers to when commentary on an event is produced in a studio while watching the event on a monitor. In the simplest operation, a commentator may create the program mix on the commentary unit itself and a program output is ready for broadcast.

Commentary/Announce Box

Commentator devices are very much part of analog PL systems and Clear-Com offers the AB-120 products for this application within the Encore partyline product family.

Commentators connect their headphones and microphone to the commentary/announce box. In-the-ear earpieces are the most commonly used listening device but communication headsets with high audio quality boom microphones are often used for sportscaster applications.

These devices allow announcers to manage their own microphone's on/off button and monitor a variety of sources. This on/off button is typically called a 'Cough Button' and can be configured as a momentary action button that turns off the microphone input only when it is held down. In addition, a commentary/announce box will feature a Co-Ordination (Co-Ord) or TB (Talkback) Circuit. This is the main off air communication channel between the commentators, the engineer, sometimes the audio A1, and the coordinator. The coordinator is an editorial position responsible for the commentators, usually a producer or director. Line level outputs are provided, often with a mixed output of all mics.

For applications that require complex reporting, commentator user stations with multiple input headphone mixers are used. Some units provide up to two or three commentary positions allowing them to set up individual headset mixes. In these applications, a commentator may receive the program audio feed as broadcast, the sound of the stadium crowd, advice on mic levels or popping from the engineer, a local input from a sports statistician and of course off air instruction from his coordinator/producer. Most units are capable of split-ear operation, individual level and pan controls for each source and 4-wire connections to intercom systems and external line devices.

2 GENERAL ANALOG PL CABLE CONSIDERATIONS

When thinking through how to install and wire any partyline intercom system, several factors must be considered. These include the number of stations, the length of the cable runs and whether single or multiple channels are required. In analog systems where multi-channel stations are connected with multi-pair cables, then crosstalk becomes an important issue. Crosstalk is not a factor with single-channel systems or multi-channel systems where each channel is run on its own individual cable to single-channel Remote Stations. While the physical considerations include ease of installation, type of cabling, station location, etc., the electrical considerations are concerned primarily with the capacitance between conductors on the intercom line, and the DC resistance in the ground return of the intercom line. More on this follows in Section 5.7 Recommendations for Various Wire Types.

> **Note:** ***A general rule with all analog PL intercoms is to never connect PIN 1 to the shell of the XLR connector.***

Excessive resistance in the conductors of the cable results in a loss of sidetone null at Remote Stations, and some overall loss of level. Excessive resistance in the ground conductor or shield greatly increases crosstalk between channels. This can significantly affect the performance of multi-channel systems.

The Clear-Com intercom line is intended to run on a shielded twisted pair cable per channel of intercom. One conductor carries full duplex ("two-way") audio, the other conductor carries the DC power for remote stations. The shield is used for ground return for audio and power. When choosing interconnect cable, keep the following considerations in mind:

a) DC resistance of the ground or common conductor affects crosstalk. For runs longer than 500 feet do not use wire smaller than 20 gauge. The total resistance of the ground return (the combined parallel sum of all shields to a location) to any point in the system should be under 1.5 ohms.

b) The capacitance of the interconnect cable affects system frequency response and side-tone stability. Total capacitance should not be greater than 0.25uF (capacitance between conductor and shield) equivalent to an intercom system containing 5000 feet of cable at 50pF per foot.

High Frequency Loss on a 200 Ω line (CC)

.05 µF = 50,000 PF	- 0.4 dB	- 0.75 dB
.1 µF = 100,000 PF	- 1.4 dB	- 2.5 dB
.2 µF = 200,000 PF	- 4 dB	- 6 dB
.3 µF = 300,000 PF	- 6.6 dB	- 9 dB
.4 µF = 400,000 PF	- 8.6 dB	- 11.3 dB
.5 µF = 500,000 PF	- 10 dB	- 13 dB

Total Capacitance includes ALL cables connected to a channel

Table 1: HF Loss to Capacitance

2.1 Cable Installation influences on Cross-Talk

When multiple channels are fed to remote stations, the amount of cross-talk between channels is proportional to the amount of DC resistance in the ground return path back to the termination of the channels.

The terminated end of the cable will not contain the cross-talk. The remote end of the cable will have the cross-talk as it is generated in the ground return and not the audio line.

The ideal installation would have all multi-channel cables originate from a central point. The termination for each channel is located at this central point. This type of installation is known as a "Star" type system. If no single multi-channel leg of the "Star" arrangement exceeds 500 ft. the crosstalk should be negligible.

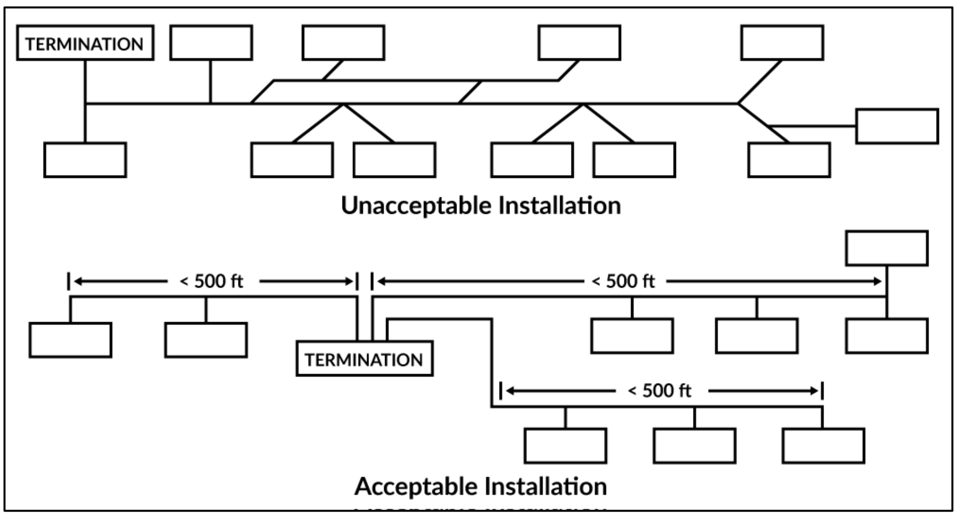


Figure 15: Central Point for termination

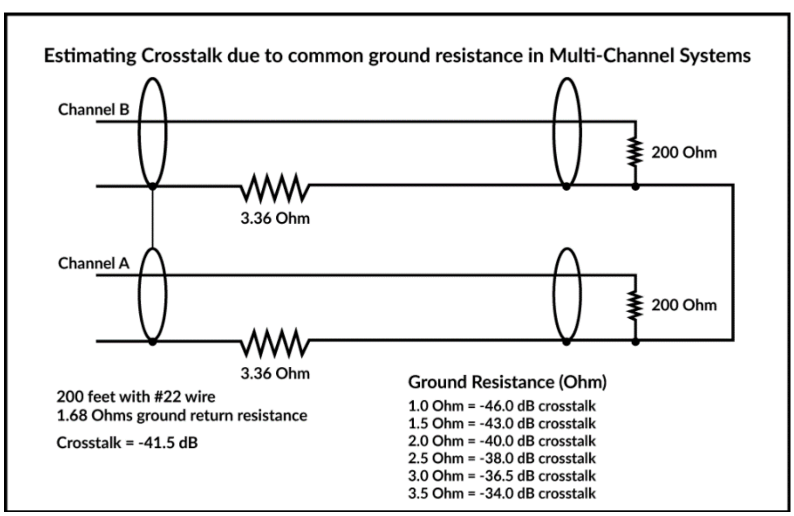


Figure 16: Central Point for termination

2.2 Cross-Talk Considerations in a Multi-Channel System

In a multi-channel system where multiple channels are run from a Main Station to a Remote Station as in the Figure 17 diagram, crosstalk can be an issue. This is because the channels will share a common ground at both ends of the cable run.

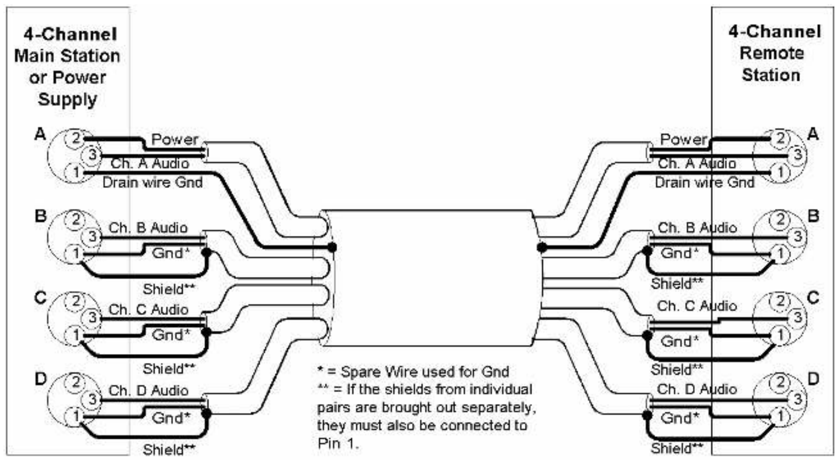


Figure 17: Party Line Multi-Channel Configuration

When multiple channels are fed to Remote Stations, the amount of crosstalk between channels is proportional to the DC resistance of the ground return path back to the channel terminations. To minimize this crosstalk between channels when running more than one channel in a multi-pair cable, keep the DC resistance of the ground return as low as possible. Ideally, this should be less than 2Ω. This can be achieved as follows:

- Keep cable runs under 500 feet. If a longer cable run is unavoidable and approaches 1,000 feet or more, make sure the appropriate lone line option switches or jumpers are set in the stations. Refer to the individual station manuals for further information.
- Use a cable whose common shield has a low DC resistance.
- Connect unused cable wires of a multi-pair cable to the Pin 1 shield.

Note: All multi-pair cables must have individually shielded pairs.

The performance of a Clear-Com system depends upon the use of Clear-Com or Clear-Com compatible headsets. Use of headsets other than these can induce crosstalk into a multi-channel system through the headset cable. Clear-Com also recommends against the use of headset extension cables or headset "Y" cables, as they will increase crosstalk in a multi-channel system.

2.3 Crosstalk Through a Mutual Capacitance of Two Conductors

A factor influenced by cable size is the *high frequency response* of the whole system affected by the *capacitance of the cable*. Capacitance occurs between any two conductors separated by an insulator.

Capacitance (taken from AV Cable Installer Guide by Steve Lampen – used with permission from the author): When two pieces of metal are separated by a nonconductor, an electrical device called a capacitor is formed. A capacitor stores voltage. The effect is called capacitance and is measured in farads. In the case of two metal wires separated by plastic, the capacitance is measured in picofarads (pF), or trillionths of a farad. How can such a small amount of anything have an effect? The reason is that the capacitance increases along the length of the cable. For instance, a 1,000-ft cable with 30 pF/ft will have a total capacitance of 30,000 pF, a significant amount.

Capacitance has a negative impact on system performance which needs to be factored into system design. Even though the capacitance is always the same on a given length of cable, it reacts to the frequency on the wire; this effect is called capacitive reactance. This effect opposes current flow, so it is also measured in ohms (Ω), like resistance. Resistance affects all frequencies equally, whereas capacitance affects high frequencies more than low frequencies. All cables have two or more conductors, so all cables have capacitance. One can substitute a different plastic with a lower dielectric constant and get lower capacitance or move conductors farther apart to reduce capacitance.

For use in intercom, a loss of 3dB at 3kHz is considered the maximum acceptable amount of impact to maintain intelligibility, though this may not be adequate in high-noise applications.

In the following example, we have assumed a 3dB loss @ 3kHz.
To determine the maximum tolerable cable length in a system it is essential to know the capacitance of the cable. In the following example, we have assumed a cable capacitance of 100pF/meter or approximately 30.5pF/foot. Here's the math:
The key formula is: *Z = 1 /2πFC*
Where Z = Ohms, F = Hertz, and C = Farads
Therefore:
C = 1 /2π FZ   For a 3kHz cut off (-6dB point) in 200Ω circuit:
C = 1 /(2x3.14) x 3000 x 200 (Ohm)
C = 1 /3768000 F or C = 1 /3.768uF or C = 0.264uF or C = 264000 pF
Permissible cable length is 264000 ÷ 100pF = 2640 meters
Permissible cable length is 264000 ÷ 30.5pF = 8655 ft.
This rule applies only to capacitance.

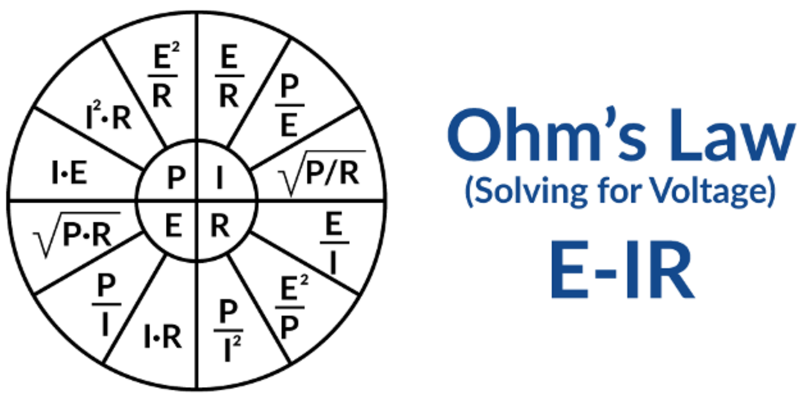
Two conductors such as a twisted pair can accumulate a large mutual capacitance over long distances. Another calculation solving for crosstalk level using the same figure of 100pF (pico-farads) per meter and a distance of 1 kilometer shows the following. The key formula again is: *Z = 1 /2πFC* which results in a total capacitance of 100nF (nano-farads) or 0.1mF (microfarad). The reactance of 0.1mF at 3000 hertz is roughly 530Ω. Referred to the system impedance of 200Ω, the apparent crosstalk is 20log (200/530) or about -8.5dB. Separating the two channel conductors by a shield greatly reduces the capacitive crosstalk so that the resistive crosstalk discussed above dominates.

Crosstalk and Starquad cable
As established above, the cable with the lowest capacitance will have the highest performance. Therefore, for intercoms in general NEVER use starquad cable as it has by definition the worst performance for these applications based on its unique design. Because you only need two wires for a balanced line, with starquad cable you combine conductors to make a pair. Most starquad cables are color coded, so you can simply combine the conductors of the same color insulation. E.g., if you have two blue wires and two white wires, you combine the blues into one wire and the two whites into one wire. Because you are combining conductors, you are also combining the capacitance. Therefore, capacitance on starquad cable is high, sometimes more than 50 pF/ft.

2.4 Rule of Thumb for Voltage Drop

The rule for analog PL intercom cabling is to minimize voltage loss to 10Volts.

For this we use a standard Ohm's Law formula:



Two pieces of information are needed-
1) I = The current source of our load, i.e. the current that a user station (beltpack, etc.) will draw from the circuit. The figure to use is the maximum current draw or the specification shown with the talk latched and signaling applied.

For our example we will use the specification for a 2 channel remote station such as the RM-702
Power
Input Voltage Range: 20-30 VDC
Input Current (Idle): <= 90mA
Input Current (Max): <=110mA

2) R = The nominal conductor resistance per 1,000 feet of the cable. This specification is available from wire manufacturers. For this example, we are using an 18AWG shielded single pair with a spec stated by the manufacturer as 6.5Ω/1,000ft.

Looking towards an application of a 100-foot run of 18AWG cable we solve for line resistance: *(Wire resistance per 1000‘/1000) X (distance in feet) of run X 2.*

Example:

18AWG = 6.5 Ohms
Formula = *6.5Ω/1000 X 100 X 2*
6.5Ω/1000 = .0065
.0065 x 100 (feet run) = .65
.65 x 2 = ***1.3Ω*** wire resistance for 100 feet of 18AWG wire.

Looking towards an application where we use 10 Remote Stations we solve for current per drop:
Number of user stations X Power Requirements
Example:

10 RM702 @ 110mA (.11A) ea.
10 X .11 = ***1.1A***

Standard Ohm’s Law formula: *E = IR*

E = I (1.1A) x R (1.3Ω) = ***1.43V Loss***

For ease of wiring deployment, and because they normally follow the same path, people have often chosen to use the multicore cable microphone ”snake” for the intercom as well as for their audio transport from stage to mixing console (and back). The problem with these for intercom is that the cable is often undersized being 24AWG or smaller and the nominal conductor resistance per 1000feet of the cable is very high.

In the example below we look at the voltage loss calculated with this type cable keeping the user station type and quantity the same.

Looking towards an application of a 100-foot run of 22AWG cable we solve for line resistance: *(Wire resistance per 1000‘/1000) X (distance in feet) of run X 2*

Example:

AWG = 14.3 Ohms
Formula = *14.3Ω/1000 X 100 X 2*
14.3Ω/1000 = .0143
.0143 x 100 (feet run) = 1.43
1.43 x 2 = ***2.86Ω*** wire resistance for 100 feet of 22AWG wire.

Looking towards an application where we use 10 Remote Stations we solve for current per drop:

Number of user stations X Power Requirements
Example

10 RM702 @ 110mA (.11A) ea.
10 X .11 = ***1.1A***

Standard Ohm's Law formula: *E = IR*

E = I (1.1A) x R (2.86Ω) = ***3.15V Loss***

Refer to Section 5.1 <u>Analog & Digital Partyline Intercom Cabling Comparison</u> below to understand how to apply the voltage/resistance to power delivery length limits.

3 CLEAR-COM DIGITAL PARTYLINE SYSTEMS

3.1 Introduction to Digital Partyline

The HelixNet system transports the intercom audio and signaling as packetized Ethernet using CSMA/CD (carrier sense multiple access with collision detection). In its base configuration, the HelixNet system supports powerline technology, where the Ethernet packets are modulated on to the powerline. This makes it possible for several stations connected to the same physical medium to share it.

Carrier sense multiple access (CSMA) is a media access control (MAC) protocol in which a node verifies the absence of other traffic before transmitting on a shared transmission medium, such as an electrical bus.

'*Carrier sense'* means that a transmitter attempts to determine whether another transmission is in progress before initiating a transmission. That is, it tries to detect the presence of a carrier signal from another node before attempting to transmit. If a carrier is sensed, the node waits for the transmission in progress to end before initiating its own transmission. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk”.

'Multiple access' means that multiple nodes may send and receive on the medium. Transmissions by one node are generally received by all other nodes connected to the medium.

In this powerline based intercom, we use Frequency Modulation to separate our audio signal from the power signal. Here the digitized audio signal is modulated with a high frequency carrier and then the modulated signal is injected into the powerline through a high pass coupler circuit. The reason for choosing the Frequency Modulation is that it gives excellent quality signal output and free from noise.

This allows Ethernet based devices such as beltpacks, speaker stations and remote master stations to be connected to the powered output using standard twisted pair shielded, 3-pin XLR, cables using the same topologies utilized in analog Partyline systems, including passive splits, daisy chains and extended cable distances. Digital Partyline systems offer improved audio quality, resiliency to external interference that the unbalanced partyline systems are susceptible to, as well as minimal crosstalk between adjacent channels.

Clear-Com Digital PL Powerline and other PL System cautions

The user should note that the HelixNet powered platform called 'powerline', and any analog or other digital partyline intercoms are dissimilar technologies and cannot be directly connected, i.e. one cannot, and should not, connect any other device except another HelixNet device on to a HelixNet powerline circuit. Doing so will cause damage to some analog devices such as TW intercom products and older Clear-Com analog PL products pre-RS-700 Series. Interfacing to analog PL systems is only possible with a purpose-built HelixNet 2-wire interface module required for installation in to the main station. See Section 3.4 on Interface Modules.

3.2 History

When HelixNet was first introduced, the Main Station provided four channels of full-duplex intercom communication, plus a separate program audio feed, to a combination of at least 20 beltpacks. At that time, wall, desktop, and remote stations were not introduced and the system did not have the capability for user stations to connect via IP network infrastructures using standard switches but only connected user stations to the powered output called "powerline".

There were 2 basic goals associated with the development of HelixNet:

1. Make it familiar
2. Make it useable with existing infrastructure

The most important attribute to the HelixNet platform is the familiarity for previous users of analog partyline. If you have read the analog chapters and have worked with Clear-Com analog PL products the layout and functions with HelixNet are almost identical with some very nice additions such as visual displays to channels and menus for ease of configurations.

So, similar to its analog counterpart, a Main Station user will have individual talk, listen, and call pushbuttons for each channel. Each channel has individual listen level controls, remote mic-kill switch to turn off all latched talks, a rear panel jack providing a buffered un-switched output of the selected microphone to be used in an external IFB system and an all-talk button access to all channels.

All Main and Remote Station have the ability for Interrupt Fold Back, (IFB) of program feeds to intercom channels on talk or call and all Main and remote Stations have extra features for special tasks such as main program feed to the stage announce output with interrupt relay closures.

Today, the number of partyline channels created in the system is 12 with option to increase this to 24 partyline channels thru the purchase of an upgrade license. Within today's HelixNet architecture, there are two basic approaches for connecting components together. All products are capable of connecting to the powered output called "powerline" created within a main station comparable to analog PL as well as on standard IT networks using commercial off-the-shelf, (COTS) IT hardware running IP protocols. All user stations can be connected in a combination of powerline and Ethernet using PoE for hybrid operation.

3.3 Stations

The HelixNet intercom products include:

HMS-4X	HelixNet Main Station
HRM-4X	4-Channel HelixNet Remote Station

HKB-2X 4-Channel (Two Display with Shift Page), HelixNet Speaker Station
HBP-2X 2-Channel HelixNet Beltpack
HXII-BP 2-Channel HelixNet Beltpack includes IP and Powerline
S-Mount Mounting Kit for HKB-2X Speaker Station

Intercom and audio interfaces include:
HLI-2W2 2-Wire Interface Module
HLI-4W2 4-Wire Interface Module

IP interfaces include:
HLI-ET2 Ethernet Linking Module
HLI-FBS Fiber Linking Module.

Networking will be discussed subsequent section.

3.3.1 HMS-4X

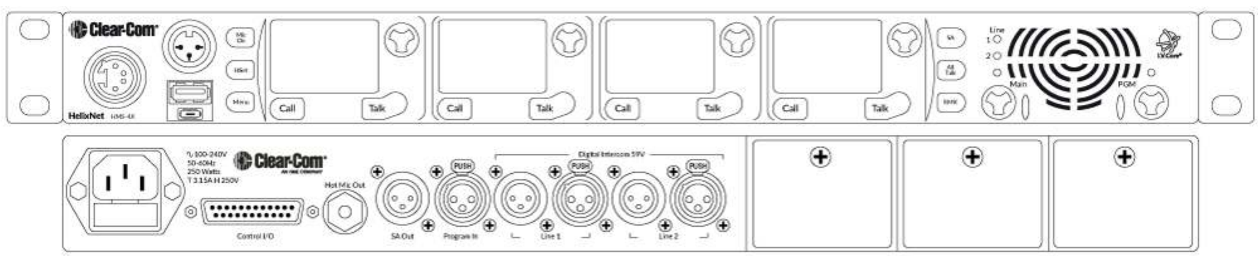


Figure 18: HMS-4X Front/Rear

Just like the analog partyline equipment, the Main Station is designed for applications in which a person needs to communicate with other people or groups of people, either individually or simultaneously, while still maintaining isolation between each of the channels. This station is usually assigned to a director, producer, stage manager or show caller depending on the workflow. The Main Station is self-powered, and does not have a power switch, button or key. The system powers up when you connect the power supply and provides power to remote Main Stations, speaker stations or beltpack headset stations if needed.

The HMS-4X intercom Main Station has two 'powerline' circuits to supply power to Remote Stations, with each circuit providing 59VDC, (i.e.±30VDC), on the line. There are two pairs of rear-panel-mounted male and female XLR3MF connectors for this purpose, designated "Digital Intercom (59V)", "Line 1" and "Line 2". It is possible to make a redundant connection on the powered intercom line between the Main Station and the Remote Stations by returning a cable from the end of the daisy-chain to the second XLR3 connector in each digital intercom pair.

The intercom Main Station connects to these stations using various cable types including shielded single-twisted-pair cables of various AWG, shielded category data cable, and coaxial cable. Y-split and star-split cable configurations can be made without the use of active splitter devices.

Pin	Function
Pin 1	Ground
Pin 2	+30VDC plus Audio
Pin 3	-30VDC plus Audio

Table 2: HelixNet Line 1/Line 2 Powerline XLR pin out

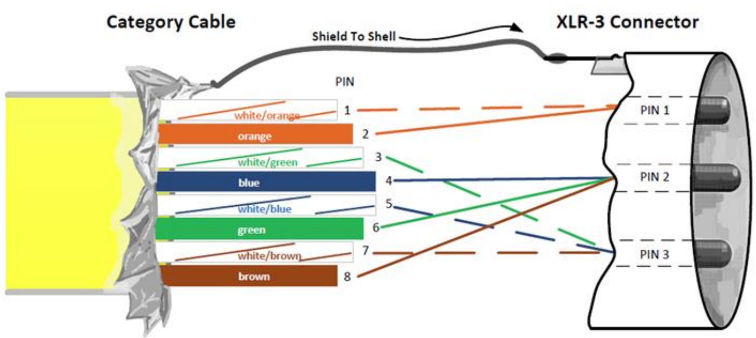


Figure 19: CAT cable to XLR3 wiring

Cat5/ 5e/ 6	XLR
White/ Orange	1
Orange	1
White/ Green	3
Blue	2
White/ Blue	3
Green	2
White / Brown	3
Brown	2
Shield/ Drain	1

Table 3: CAT cable to XLR3

HelixNet powerline can be deployed using a wide range of topologies, both complex and straightforward. The following table describes three standard types of topology.

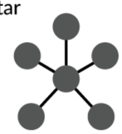
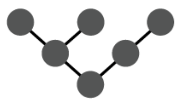
Topology	Description
Daisy-Chain 	In a daisy-chain topology, the Main Station is connected to the first unit. The units are then connected in a series, using the pass-through connector on the back/rear of the device to pass on the connection. No termination is required. HXII-BP-X4 beltpacks require a passive Y splitter for daisy chaining. Daisy-chains can be either linear or loop back to the Main Station to form a ring for redundancy.
Star 	In a star topology, the Main Station is connected to a passive (Y) splitter such as an XLR Bulkhead or Mult-box. The units are then connected to the splitter point-to-point.
Tree 	A tree topology is a more complex version of the star topology, using serially connected passive (Y) splitters. The units are connected to each splitter point-to-point (forming a branch of the tree).

Table 4: Topology types

The HMS-4X Main Station includes a front-panel Menu button to place the Main Station in Menu mode. Menu mode is used to configure the settings for the Main Station and system-wide parameters including channel and audio settings, system administration, system performance monitoring and diagnostics, configuration saves or uploads, and software updates. The four OLED displays enables the programming of Main Station parameters when the front-panel Menu button is pressed, acting in conjunction with each other in a flat menu structure so that all parameter selections are available across the displays. The menu hierarchy proceeds left to right.

What is a Keyset?

A Keyset is a set of controls associated with an audio assignment. The Main Station and Remote Station has 4 audio assignments (Keysets), and the speaker station and beltpack has two main audio assignments (Keysets).
The Keyset on the Main Station and Remote Station is made up of four (4) viewing screens, each with an associated rotary encoder controller, a Call key and a Talk key. These controls display and regulate the audio routes associated with the Main Station. The viewing screens display menu options and diagnostics as well as controlling audio assignments.

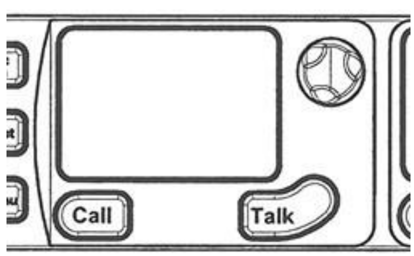


Figure 20: HelixNet Main /Remote Station Keyset example

Three rear-panel slots are provided to accommodate any mix of 2-wire, 4-wire, copper Ethernet, or optical Ethernet interface modules. These interface modules allow the Main Station to connect with a variety of digital matrix, wireless, or analog wired party-line intercom systems, as well as other audio communication circuits. Ethernet connectivity is described in the 'network' section below. Rear panel Control I/O consisting of a DB-25F connector to provide four relay circuits, individually selectable to be normally open or normally closed, and four opto inputs.

Pin	Function	Pin	Function
Pin 1	Relay 1 NC	Pin 14	Relay 1 - Pole
Pin 2	Relay 1 NO	Pin 15	Relay 2 NC
Pin 3	Relay 2 - Pole	Pin 16	Relay 2 NO
Pin 4	Relay 3 NC	Pin 17	Relay 3 - Pole
Pin 5	Relay 3 NO	Pin 18	Relay 4 NC
Pin 6	Relay 4 - Pole	Pin 19	Realy 4 NO
Pin 7		Pin 20	+5V
Pin 8	GND	Pin 21	+5V
Pin 9	GND	Pin 22	Opto 1-
Pin 10	Opto 1+	Pin 23	Opto 2-
Pin 11	Opto 2+	Pin 24	Opto 3-
Pin 12	Opto 3+	Pin 25	Opto 4-
Pin 13	Opto 4+		

Table 5: Control 25way female D-type I/O pin out

The HMS-4X Main Station connectors consist of the following:

- Front Panel: Headset, 4-pin XLR–M
- Front Panel: Gooseneck microphone, 3-pin Tuchel connector
- Rear Panel: Intercom Line, Two, (2) set 3-pin XLRMF loop-thru
- Rear Panel: Hot Mic /IFB Interface, 1/4" (0.64 cm) phone jack
- Rear Panel: One XLR3F, providing one audio line level input labeled Program
- Rear Panel: One XLR3M, providing one audio line level output labeled SA Out for Stage Announce or recording purposes
- Rear Panel: DB25 providing four (4), opto inputs, four (4), relay contact closures as delineated above.

Linking

Up to three Main Stations may be linked. In earlier versions of HelixNet (2.0 and below) linking Main Stations was a way of expanding the Channel capacity of your system (4 Channels on each Main Station). In HelixNet 3.0 and later, this is no longer necessary as each Main Station already has 12 Channels, with the option of licensing another 12 (24 Channels in total). A system, regardless of the number of linked HMS units, will have 12 Channels by default with the potential of increasing that number to 24 with the purchase of a license for each HMS within the **Link-Group**.
Linking Main Stations in HelixNet has the following benefits:

- More beltpack connections (20 per Main Station)
- System distribution as far as your LAN allows.
- The ability to configure all devices from an Internet browser using the CCM (Core Configuration Manager) see below- or from the device menus.
- Using a role-based setup.
- Expanding a Main Station from four keys to 24 key using the expansion key mode. (see below)

Main Stations can be linked using different methods:
- By Ethernet and RJ45 cable in your LAN.
- By fiber cabling between units.

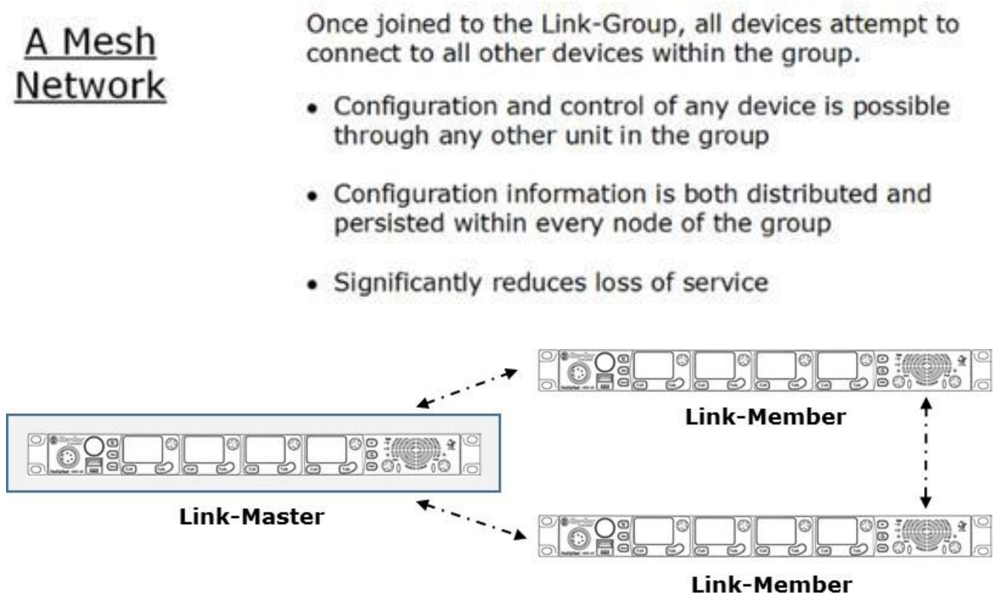


Figure 21: Example Mesh System Topology

> ***Note:*** *Linked Main Stations should have the same number of licensed channels. If you link a 24 channel device to a 12-channel device the system will default to 12 channels only.*

Key Expansion Groups
HelixNet units can be expanded and stacked to allow for visual monitoring and easy control of multiple channels. Up to five devices (HMS-4X, HRM-4X) can be stacked in an Expansion Group, allowing for convenient observation and control of up to 24 Keysets. This gets rid of the need for multiple key presses and microphones. When using this type of connectivity, the Expanded panels act as slaves to the Host panel.

Refer to a HelixNet User Guide for further **Link-Group** setup information.

The Core Configuration Manager (CCM)

The Core Configuration Manager (CCM) interface provides an intuitive software utility for the Helixnet platform and other IP Clear-Com products on any browser-enabled platform. The CCM facilitates a quick and simple means of configuring any devices in a Link-Group, including role-based configuration of endpoints, account management, save and restore, and live monitoring of all system components.

Although there are subtle differences within CCM when directed towards FreeSpeak II, LQ Series and HelixNet, the common design and workflow of the interface allows a system administrator to quickly navigate from one product line to another without the need for the training on different configuration tools. The pages logically group similar components and/or behaviors to maximize workflow efficiency.

Key Features

- Free browser-based configuration tool
- Supports latest versions of all major web browsers
- Common look and feel across FreeSpeak II, LQ Series and HelixNet systems
- Configuration for any device within Link-Group
- Multiple administrator monitoring and configuration of Link-Group
- Overview of all system components and interfaces on a single page
- Dark theme for low-light environments
- Password protected

HBP-2X Beltpack

The HBP-2X intercom beltpack only communicates with the Main Station and other connected remote intercom stations using the powered powerline output. It can be connected with the Main Station and other user stations using various cable types, including shielded single-twisted-pair cables of various AWG and shielded category data cable wired to XLR3 connectors. The beltpack receives audio data, control data, and DC power via the same cable.

The beltpack can receive any of the system channels of intercom communications simultaneously over the single cable, and can select among, monitor, and communicate on any two of those channels at any given time, and receive and monitor a separate level-controllable program audio feed. Using the beltpacks menu function, the user can independently select among and change channel assignments as necessary from any position on the intercom line.

The beltpack includes controls and display capabilities to access and communicate on at least two simultaneous intercom channels, including:

- OLED display, showing the name and relative level of each selected channel
- Momentary/latching Talk buttons
- Call buttons
- Independent PL Rotary listen level controls
- Independent Program Rotary listen level control

A separate rotary level control for program is provided, and when that level is adjusted the OLED display shall temporarily show the relative level of the program signal, and then return to the standard PL screen.

The beltpack provides:

- Menu-selectable responses to a received Call, including vibrating and visual alerts
- Brightness level of the OLED screen
- Variable, and controllable Talk, and Call buttons
- Sidetone level can be adjustable in four 6-dB steps, ranging from 0 dB to -18 dB
- Headphone peak level is adjustable in four 6-dB steps, ranging from +6 dB to -12 dB

The menu is accessed via a front-panel push button, with parameter selection and scrolling controlled by the left and right Call buttons and rotary channel level controls. The beltpack menu displays the existing firmware version, and its firmware is automatically updated via the Main Station to the system's current version when connected. The beltpack has a micro-USB connector which can be used for a micro-USB flasher or to update firmware.

There is a pair of 3-pin XLR intercom connectors (1 x XLR3F, 1 x XLR3M), on the bottom beltpack panel wired as pass-through so that it may be connected to the intercom line and then connected to the next Remote Station or looped-back for redundancy. A four-pin XLRM intercom headset connector is provided, with a five-pin connector optionally available. A 2.5-mm TRS connector is provided for compatible small-format headsets for listen only.

Figure 22: HelixNet Beltpacks

HXII-BP Beltpack

The HXII-BP intercom beltpack communicates with the system using the powerline output as well as using IP protocols on an IT network thru switches providing PoE.

When connected with the Main Station, and other user stations using the powered 'powerline' output, various cable types including shielded single-twisted-pair cables of various AWG and shielded category data cable wired to 3pin XLR connectors can be used. The beltpack receives audio data, control data, and DC power via the same cable when connected to powerline. Connection to a Main Station over an IP network requires an Ethernet interface module, (HLI-ET2), fitted to the extension bay of the HMS-4X Main Station. See section 4.2 for networking with HLI-ET2. The beltpack is connected to a port of a network switch providing power over Ethernet (PoE).

Note: *The HXII-BP beltpack must be paired to the Main Station.*

Beltpack Keysets

The HXII-BP-4X beltpack includes keyset controls and display capabilities to access and communicate on two simultaneous intercom channels, including:

- OLED display, showing the name and relative level of each selected channel
- Momentary/latching Talk buttons
- Call buttons
- Independent PL Rotary listen level controls
- Independent Program listen level controls

  The listen level (volume) of the Program Feed to the beltpack is adjusted using up and down control buttons on the front of the beltpack.

Figure 23: HelixNet Beltpack Keyset example

The HXII-BP-4X beltpack includes all of the features and functions and maintains the same feel as the older HBP-2x beltpack with the following exclusions/additions:

- There is no loop-thru XLR connector for the powerline circuit therefore a simple male -to- 1 female/1 male "Y" cord XLR adapter is required.
- There is no 2.5-mm TRS connector provided for compatible small-format headsets for listen only.
- The beltpack includes a EtherCon connector to facilitate connection to a network switch port with PoE.

3.3.2 HRM-4X HelixNet Digital Intercom Four-Channel Remote Station

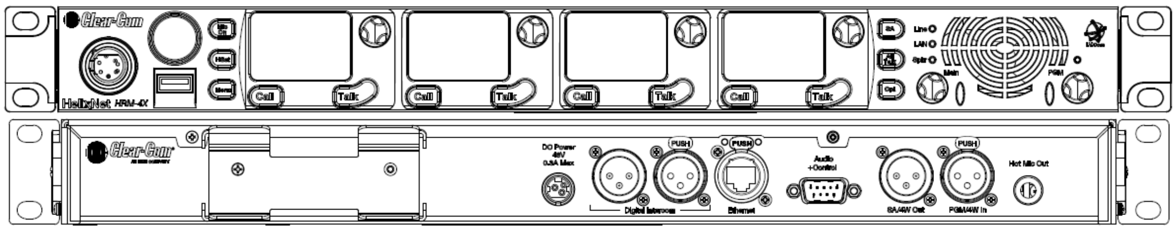


Figure 24: HelixNet HRM-4X Remote Station Front/Rear

The HRM-4X Remote Station (see Figure 24), communicates with the system using the powerline output as well as using IP protocols on an IT network thru switches providing PoE. When connected with the Main Station, and other user stations using the powered powerline output, various cable types including shielded single-twisted-pair cables of various AWG and shielded category data cable wired to 3-pin XLR connectors can be used. The HRM-4X receives audio data, control data, and DC power via the same cable and additionally has the capability to be locally powered via an external power supply or a Power-over-Ethernet (PoE) connection. Connection to a Main Station over an IP network requires an Ethernet interface module, (HLI-ET2), fitted to the extension bay of the HMS-4X Main Station.

The rack-mountable station provides access to four simultaneous channels of intercom communication, plus program audio. The front panel has separate keyset controls for the four communication channels, each including:

- OLED display, showing the name and relative level of each selected channel
- Momentary/latching Talk buttons
- Call buttons
- Independent PL Rotary listen level controls
- Independent Program listen level control

The front panel includes a speaker with a rotary encoder level control and multi-color LED level indicator. The station has separate buttons to activate the headset or the gooseneck mic.

The HRM-4X Remote Station includes a front-panel Menu button to place it in Menu mode. The four OLED displays act in conjunction with each other in a flat menu structure so that all parameter selections are available across the displays. The menu hierarchy proceeds left to right. Menu mode is used to configure the settings for the Remote Station only including audio, channel, control I/O, station, network, administration and diagnostic settings. The menu button is programmable and may be locked if access is not required.

The HRM-4X Remote Station connectors consist of the following:

- Front Panel: Headset, 4-pin XLR–M
- Front Panel: Gooseneck microphone, 3-pin Tuchel connector
- Rear Panel: Intercom Line, one set 3-pin XLRMF loop-thru
- Rear Panel: Hot Mic /IFB Interface, 1/4" (0.64 cm) phone jack
- Rear Panel: One DB9, providing one opto input, one relay contact closure, and one auxiliary audio balanced input and output
- Rear Panel: One XLR3F, providing analog line level audio input labeled for PGM
- Rear Panel: One XLR3M, providing analog line level output used for SA
- Rear Panel: One EtherCON RJ-45, providing Ethernet connection to Main station or third-party PoE device
- Rear Panel: 3 pin Mini-DIN connector for external power supply

Note: *The DB9 audio circuits are wired in parallel to the 3-pin XLR PGM input and SA output and cannot be used as a second source/destination.*

3.3.3 HKB-2X HelixNet Digital Intercom Four-Channel Wall or Desktop Speaker Station

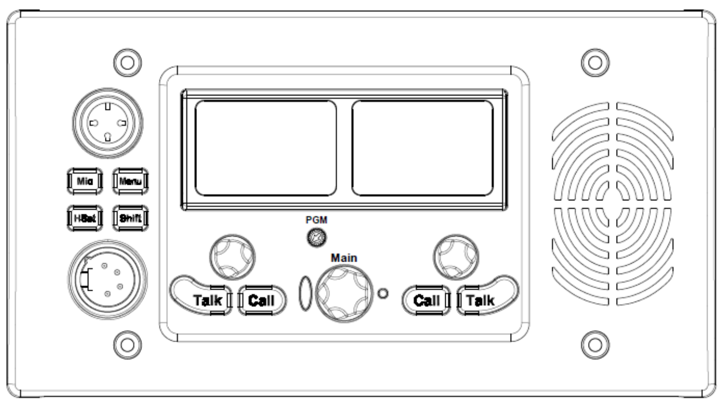


Figure 25: HelixNet HKB-4X Speaker Station

The HKB-2X Remote Speaker Station (see Figure 25) communicates with the system using the powerline output as well as using IP protocols on an IT network thru switches providing PoE. When connected with the Main Station, and other user stations using the powered powerline output, various cable types including shielded single-twisted-pair cables of various AWG and shielded category data cable wired to 3-pin XLR connectors can be used. The HKB-2X receives audio data, control data, and DC power via the same cable and additionally has the capability to be locally powered via an external power supply or a Power-over-Ethernet (PoE) connection. Connection to a Main Station over an IP network requires an Ethernet interface module, (HLI-ET2), fitted to the extension bay of the HMS-4X Main Station.

The Speaker Station receives four channels of intercom communications simultaneously over the single cable, and is able to select among and communicate on any two of those channels at any given time, while monitoring all four channels. Using the Speaker Station's Shift Page button, the user can access two additional intercom channels with the ability to toggle back and forth between the two pairs of channels.

The Speaker Station provides the following keyset controls and display capabilities:

- OLED display, showing the name and relative level of each selected channel
- Momentary/latching Talk buttons
- Call buttons
- Independent PL Rotary listen level controls
- Independent Program listen level control

The Speaker Station has connections for a headset and a gooseneck microphone, with a selector button to choose between the sources. The integral loudspeaker is powered with an amplifier providing 5 watts RMS, with a frequency response of 30 Hz to 10 kHz, +/-3 dB.

Sidetone level is adjustable in four 6-dB steps, ranging from 0 dB to -18 dB. Headphone peak level is adjustable in four 6-dB steps, ranging from +6 dB to -12 dB. The menu is accessed via a front-panel push button, with parameter selection and scrolling controlled by the left and right Call buttons and rotary channel level controls.

The Speaker Station is mountable into a standard 4-gang ANSI/NEMA OS 1 electrical box, or onto an optional enclosure to enable desktop use. Keyholes are provided in the corners of the front panel to accommodate wall mounting. When mounted in an optional enclosure, one each XLR-3M/F connector is provided allowing the Speaker Station to be daisy-chained on the powered powerline intercom line. A RJ-45 Ethernet jack is provided for connection to the Main station or to a third-party PoE device. A mini-DIN connector is included to allow a local DC power adapter.

3.4 HelixNet Audio Interface Modules

For reference, all Audio and Network interface modules that insert in the rear of the HelixNet HMS main station have the same rear edge connector.

> ***Note:*** *The interface modules are NOT hot pluggable. Ensure that the Main Station is powered down before inserting or removing modules.*

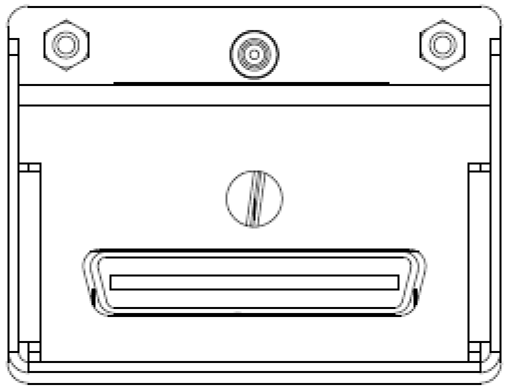


Figure 26: HelixNet module rear view edge connector

3.4.1 HLI-2W2 Two-Wire Interface Module

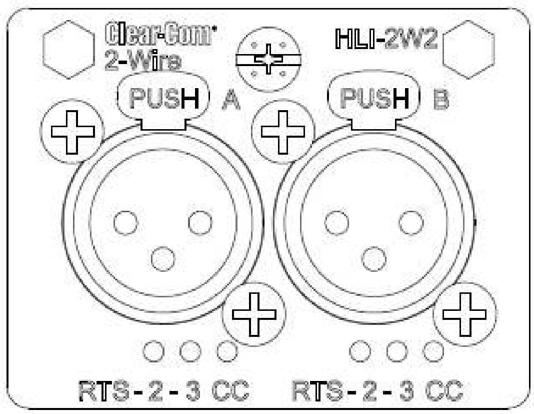


Figure 27: HelixNet HLI-2W2 module

The modular interface is used to connect two, one channel 2-wire analog audio circuits, such as a Clear-Com or RTS partyline intercom, or a wireless 2-wire intercom, to the HelixNet digital intercom system. Each of the connected external 2-wire circuits can be assignable to any HelixNet system intercom channel. The interface mounts into any of the HMS-4X Main station's three module bays, and incorporates a rear-panel edge connector to make all the necessary audio and electrical connections.

The interface offers two 3-pin XLRF connectors to connect intercom lines and are RTS /Clear-Com selectable, via the Main station's menu or browser. Specifications provide a nominal input level of -12 dBu (RTS) and -18 dBu (Clear-Com), with a maximum level before clipping of +6 dBu. Input impedance is greater than or equal to 10K ohm bridging. The audio output frequency response is 40 Hz to 10 kHz, +/-3 dB, with a noise level of -75 dBu or better, and distortion of less than 0.2% THD at 1 kHz.

> ***Note:*** *An external DC voltage range between 20 and 30 volts must be provided by connecting a system power supply.*

3.4.2 HLI-4W2 Four-Wire Interface Module

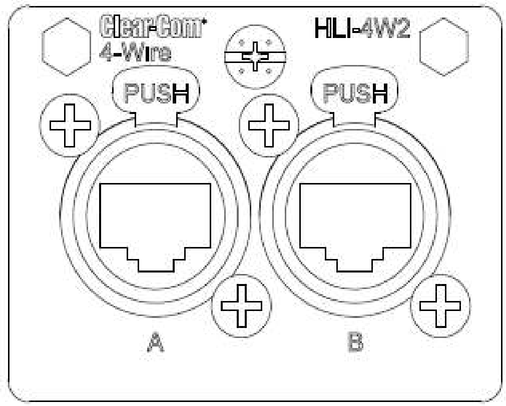


Figure 28: HelixNet HLI-4W2 module

The modular interface is used to connect two 4-wire analog line-level audio circuits to the HelixNet digital intercom system. Each of the connected external 4-wire circuits is assignable to any of the HelixNet system intercom channels. The interface mounts into any of the HMS-4X Main station three module bays, and incorporates a rear-panel edge connector to make all the necessary audio and electrical connections.

The interface includes two RJ-45 (Ethercon) connectors to connect 4-wire circuits. Specifications include a nominal input level of 0 dBu, with a maximum level before clipping of at least +18 dBu. Input impedance is greater than or equal to 10K ohm. The interface provides both transformer and electrical balancing for the incoming and outgoing audio paths to eliminate induced hum and noise.

Audio input frequency response is 20 Hz to 10 kHz, +/-3 dB, with a noise level of -65 dBu or better, and distortion of less than 0.2% THD at 1 kHz

4 NETWORKING AND DIGITAL PARTYLINE SYSTEMS

4.1 Application Background

Large theatres, live Event and remote Broadcast production customers are very interested in HelixNet Partyline because they require a solution that provides more than 20 intelligent drops (beltpacks) and many PL channels. Mid-sized venues require (on average) 5-10 partyline channels with the largest venues requiring 10-20 partylines and 60+ drops.

The vast proliferation of IP-based data networks presents an opportunity for Clear-Com to offer an even more powerful, flexible partyline communications system by taking advantage of the available telecom/data-com infrastructures.
HelixNet Partyline introduces the ability to connect stations directly or over a LAN via Ethernet. This connection provides increased node/user capacity and/or the ability to connect channels/system in disparate locations; multiple campuses, studios, venues located anywhere within a dedicated Local Area Network. HelixNet Partyline resource expansion methodology is quite logical and very similar to how intercom resources are expanded and managed in the analog partyline domain. This linking methodology not only aids in setup and configuration, it offered flexible intercom services to productions that vary in scale.

HelixNet station-to-station networking function is made possible with the HLI-ET2 Ethernet Module. Main Stations and all other Helixnet User stations can connect directly or through a LAN using conventional IT switches. The HLI-ET2 comes standard with two RJ-45 jacks. Specifications are given below. A HLI-FBS Fiber Module is also available for linking stations over long distances. The Fiber Module has two fiber ports using small form-factor pluggable, (SFP) modules for simple exchange of fiber transceivers. The standard SFP is Single-Mode with Multi-Mode offered as an option.

Multiple intercom Main Stations may be linkable via Ethernet modules or in a daisy-chain configuration with optical fiber interfacing modules, enabling the summation of 12 channel or 24-channel intercom systems. Ethernet and optical fiber interfacing modules have provision to be used concurrently in the same network. Linking HelixNet Main Stations together creates a network that pools channel resources of the individual stations. Linked main stations can dynamically discover each other thus giving HelixNet users the capability to share multiple digital partyline channels plus program inputs and any 2-wire or 4-wire interfaces in a network distributed system.

4.2 HLI-ET2 Ethernet Interface Module

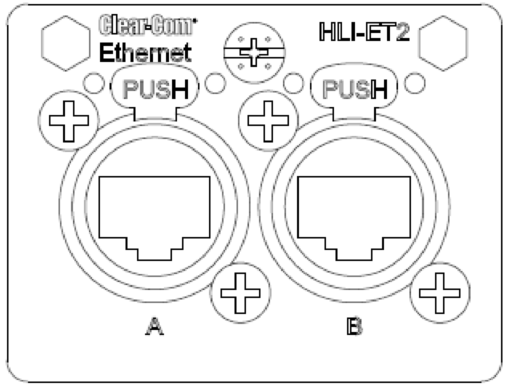


Figure 29: HelixNet HLI-ET2 module

The modular interface offers two Ethernet LAN network connections, enabling multiple HMS-4X HelixNet Main stations, Speaker and Remote stations to be connected. The connection use 10/100BaseT protocol and supports DHCP capability. Within a network the stations share all digital partyline channels, program audio signals and any or all 2-wire and/or 4-wire interface audio, and makes them available to any station or beltpack within the linked system. All channel labels are transferred across the system automatically.

The interface is equipped with two RJ-45 (Ethercon) connectors to interconnect the Main stations in to a network. The HLI-ET2 interface incorporates a rear-panel edge connector to make all the necessary audio and electrical connections and designed to be installed into any of a HMS-4X Main station three module bays.

Both ports are configured to bridge traffic from one port to the other to work in a daisy-chained configuration. Spanning Tree Protocol is not enabled on these ports, therefore do not connect them both to the same network.

The Ethernet interface module is able to operate with the fiber optic interface module (HLI-FBS) within the same network.

4.3 HLI-FBS Fiber Interface Module

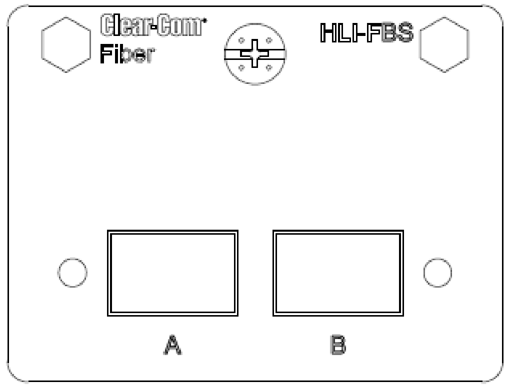


Figure 30: HelixNet HLI-FBS module

The modular interface provides for two duplex optical fiber connections, enabling multiple HMS-4X HelixNet Main stations to be connected. The connection employs duplex single-mode and multi-mode optical fiber protocol. Via the connection, the stations share all digital party-line, program audio signals and any or all 2-wire and/or 4-wire interface audio and make them available to any station or beltpack within the linked system. All channel labels are transferred across the system automatically. The fiber optic interface module is able to operate with the HLI-ET2 Ethernet interface module within the same network.

The HLI-FBS interface provides slots for one or two hot-pluggable SFP, (small form-factor pluggable) LC duplex transceivers. The interface inserts into any of the main station's three module bays, and has a rear-panel edge connector to make all the necessary audio and electrical connections.

4.4 Network Considerations

For HelixNet 3.0 and above, it is not necessary to be on the same Subnet.

The HelixNet HMS-4X, HRM-4X, HKB-2X and HXII-BP uses a 100Mb NIC.

HelixNet V3 TCP/UDP

Port 655 TCP - Linking HMS-4X
Port 80 HTML – Core Configuration Manager
Port 6000 TCP - Pairing with HKB-2X, HRM-4X, and HXII-BP
Port 6001 TCP - Authenticate, update, reboot.
Port 5353 UDP - mDNS, Expansion, Pair/Link by Name,
Port 6001 UDP - Audio

Managed Ethernet Switch

When connecting HelixNet to managed network switch ensure that the network ports are set to Auto-Negotiation or 100Mb full duplex.

If the system is using more than one switch the link between the switches should be set to auto-negotiation or both side of the link should have auto-negotiation off. One of the most common causes of performance issues on 10/100Mb Ethernet links occurs when one port on the link operates at half-duplex while the other port operates at full-duplex. This occurs when one or both ports on a link are reset and the auto-negotiation process does not result in both link partners having the same configuration. It also can occur when users reconfigure one side of a link and forget to reconfigure the other side.

Unmanaged Ethernet Switch

Since unmanaged Ethernet switches do not allow user manage port speeds, it is best to use a 10/100Mb switch.

If an unmanaged Ethernet switch is connected to a managed Ethernet switch, the port on the managed switch needs to be configured to 100Mb full duplex.

Hubs

Hubs should not be used with HelixNet, because when a hub receives a packet at one of its ports, it retransmits (repeats) the packet to all its ports. This mean bandwidth is wasted because all traffic is sent to all ports.

IP Configuration

The HelixNet HMS-4X intercom Main Station by default sets automatic IP addressing (DHCP) enabled with menu option to disable DHCP and set static IP addresses.

For this to work properly in an existing IP network there must be a DHCP server handing out IP addresses. If no DHCP server is found, a Main Station will revert to an unused link-local address in the 169.254.0.0/16 block.

A link-local address is an IP address within the local segment of any network. Routers do not pass information to these as link-local addresses and are not guaranteed to be unique beyond a single network segment. When first connected to a network, your HelixNet device will attempt to get an IP address via Dynamic Host Configuration Protocol (DHCP). If no DHCP server is available, the unit will automatically enter link-local IP mode. A link-local IP address will take the form: 169.254.xxx.xxx.

> ***Note: In link-local, the address will change each time the device reboots resulting in potential loss of connection to endpoints. The units will operate in link-local, but for optimum performance it is recommended that they are used with static network settings.***
>
> ***The preferred solution is to take a HelixNet device out of link-local mode is to disable DHCP and set static IP addresses through the Networking menu from the device front menus. Please confirm with your network administrator that there will be no IP clashes with this address.***

HelixNet Network Bandwidth

Specification	Value
Latency on Powerline	40-80ms (Depends on cable type and length, and how many devices are connected. The greater the number of devices, the greater the latency.)
Latency over IP Network	30ms + Network Latency (Main Station to Main Station)
Bandwidth used	300 kbps per active Talker, for a maximum of 1 talker per device in the system Each Beltpack and Speaker Station counts as 1 device Each Main Station and Remote Station counts as 2 devices
IP version	IPv4

Table 6 – Networking Specifications

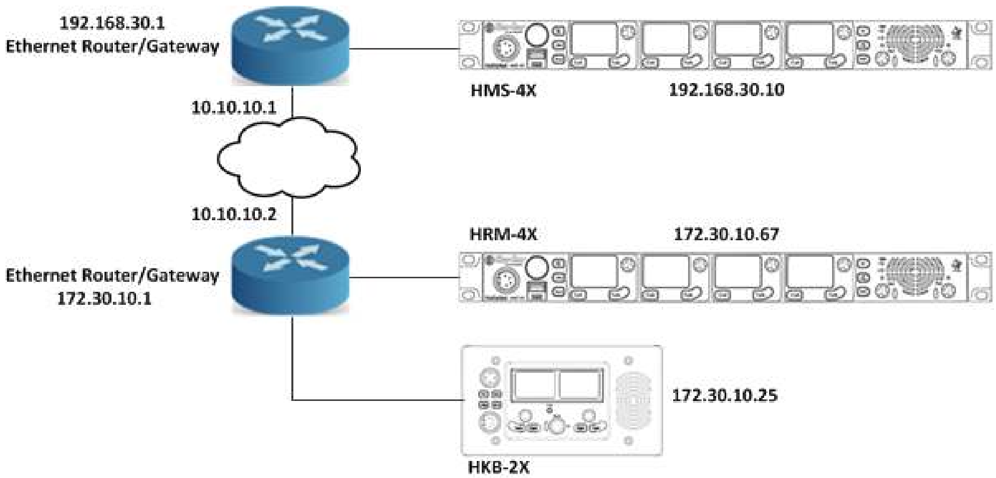
HelixNet audio is 300kbps per audio stream. The total IP bandwidth usage is proportional to the number of devices connected over IP. A detailed example of bandwidth requirement calculations follows:

For example, two linked HMS-4X, 10 HBP each, three HRM paired to HMS-A and five HXII-BP paired over IP to HMS B. Pressing All Talk on HMS A sends out 1.2Mbps from HMS-A to HMS-B plus 6Mbps to 10x HBP on HMS-A, (3Mbps) & 10x HBP on HMS-B, (3Mbps) plus 1.8Mbps over IP from HMS -A (600kbps to each HRM and 600kbps to HMS-B) and 1.5Mbps from HMS-B (300kbps to each HXII-BP)

An HMS-4X to be "reachable" from where the HRM/HKB is on port 6001 (i.e. if you connect a computer where the HRM/HKB is you should be able to "ping" the HMS address or at least get a telnet connection to port 6001) can be achieved in many ways. Here are two of the more commonly utilized examples:

Scenario A

Company networks typically have multiple subnets or VLANs. IP routers are already configured to route IP traffic across those subnets. The HMS and HRM/HKB should have their IP address, Subnet mask and Gateway set properly (all automatically done when there is a DHCP server). On the HRM/HKB deployed in a different subnet you need to Pair to Station by entering the IP address of the HMS.



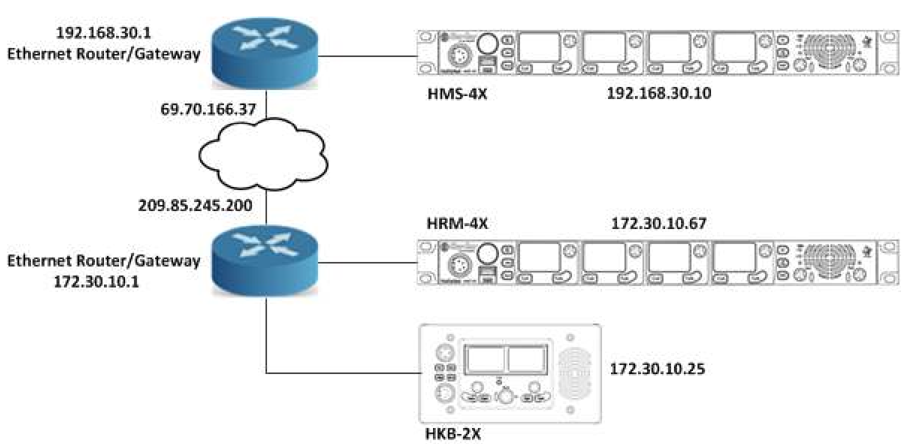
HMS IP Preferences:
IP Address: 192.168.30.10
Subnet Mask: 255.255.255.0
Gateway: 192.168.30.1

HKB IP Preferences:
IP Address: 172.30.10.25
Subnet Mask: 255.255.0.0
Gateway: 172.30.10.1
Pair to Station:192.168.30.10

HRM IP Preferences:
IP Address: 172.30.10.67
Subnet Mask: 255.255.0.0
Gateway: 172.30.10.1
Pair to Station:192.168.30.10

Scenario B

The HMS is in a private/different network, not directly reachable from where the HRM/HKB is. A reachable IP router/gateway where the HMS is must be configured to forward all the IP traffic for port 6001 TCP/UDP to the HMS:



HMS IP Preferences:
IP Address: 192.168.30.10
Subnet Mask: 255.255.255.0
Gateway: 192.168.30.1

HKB IP Preferences:
IP Address: 172.30.10.25
Subnet Mask: 255.255.0.0
Gateway: 172.30.10.1
Pair to Station:69.70.166.37

HRM IP Preferences:
IP Address: 172.30.10.67
Subnet Mask: 255.255.0.0
Gateway: 172.30.10.1
Pair to Station: 69.70.166.37

Here the Router/Gateway on the HMS side must port-forward everything coming to:6001 TCP and:6001 UDP to 192.168.30.10:6001

HelixNet pairing by Name
If a HKB-2X, HRM-4X, and HXII-BP pair to a system by name. The system uses mDNS to propagate HMS and HRM presence in a network. As a device populates it's mDNS entry, it specifies an ID, an IP address, a name and a list of services.
When configuration changes, the mDNS entry is updated and all devices connected "by name" will update and re-pair/link/expand as required.

For in depth information see the HelixNet IP Network Technical Guide at:
http://www.clearcom.com/upload/download/HelixNet-IPNetworkGuide.pdf

5 DIGITAL & ANALOG PARTYLINE INTERCOM CABLING COMPARISON

5.1 Overview

In both analog partyline systems, which have been in use since the 1960s, and more recently introduced digital partyline systems, standard 3-pin mic cable is used between a Main Station and beltpack to carry two things:

- Power
- Data (Audio and Control)

The goal of this technical chapter is to provide readers with a clear understanding of the cable characteristics important to the performance of partyline intercom systems. Wiring and crosstalk characteristics of analog partyline have been covered to some extent in previous sections.

HelixNet Digital Partyline uses wired cable infrastructure to transport audio and data over a range of frequencies. The maximum frequency used for transmission is approximately 25MHz. Depending on the amount of audio and data transmitted, HelixNet digital Partyline can work with as much as 90dB signal attenuation. However, the receivers are very sensitive and are susceptible to crosstalk between cables. Therefore, it is important to maintain cable shield integrity through all connectors, splitter boxes and patch panels.

The symptoms of crosstalk are:

- Main Station front panel LINE LED indicator turns amber or red.
- Beltpacks, Remote Stations and Speaker Stations take longer than usual to boot and connect.
- Beltpack, Remote Station and Speaker Station front panel signal strength indicators show lower than usual signal strength (zero or one bar).
- Main Station diagnostics screen (Diagnostics->Powerlines) indicates collisions or errors on the line.

5.2 Background

Each cable type has a number of physical and electrical characteristics that affect its suitability for an application. Among these are:

Number of cores

- Core twist properties
- Shielding
- Overall diameter
- Flexibility
- Voltage rating
- Flammability rating

The electrical characteristics that are of particular interest for those transitioning from analog to digital partyline systems are:

- DC resistance
- Signal attenuation

5.3 Power Limits

The one factor that affects a cable's ability to deliver DC power is its DC resistance which is related to its gauge. Figure 31 shows the relationship between cable gauge and DC resistance.

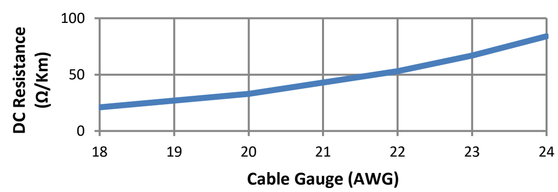


Figure 31: Cable DC Resistance vs Wire Gauge

The larger the cable (lower gauge number), the lower the DC resistance and the better its ability to deliver power over a longer distance. Here are some examples for both analog and digital partyline.

5.3.1 Analog Partyline

In an analog partyline system, the master station delivers around 30V DC to the line. The beltpack requires a minimum voltage of around 12 to 20V DC (varies dependent on type) at its input in order to operate. Most analog beltpacks draw the same current (around 25mA) from the line no matter what voltage they receive at their input.

Figure 32 shows how a cable can be considered as a simple resistance for calculation of power delivery length limits. The resistance of both cores used in the power delivery loop, the 30V core (R1) and the ground reference core (R2) must be considered in the calculation.

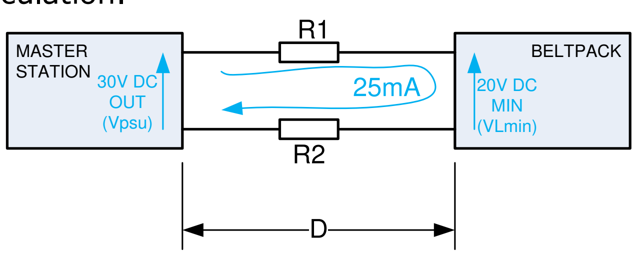


Figure 32: Analog Partyline Voltage Drops

In the example given in Figure 32, the maximum allowable value for the cable DC resistance can be calculated as follows:

$$(R1 + R2)\text{max} = \frac{V_{psu} - V_L min}{I_L} = \frac{30-20}{.025} = 400\Omega$$

Based on this calculation, assuming R1 and R2 are equal, a cable with a resistance of 33 ohms/km (such as 20awg **Belden 9463F**) could achieve a distance of:

$$Dmax = \frac{\mathrm{R_{MAX}}}{\mathrm{R/Km}} = \frac{400}{2*33} = 6Km$$

Note that this is with a single beltpack. Adding nine more to the end of the same cable, making a total of **ten beltpacks** gives a maximum power distance limit of:

$$Dmax = \frac{V_{psu} - V_L min}{I_L \text{ x R/Km}} = \frac{30-20}{0.25\ x\ 66} = \mathbf{0.6Km}$$

The nature of an analog partyline system means that when it reaches its maximum cable length for power distribution, the user will experience a lowering of maximum audio level available at headsets before distortion levels increase.

5.3.2 Digital Partyline

In a Clear-com digital partyline system, the master station delivers around 59V DC to the line. The beltpack requires a minimum of around 24VDC to operate. Clear-Com digital partyline beltpacks take the same amount of power (around 3.5W) from the line no matter what voltage (within a specified range) they receive at their input.

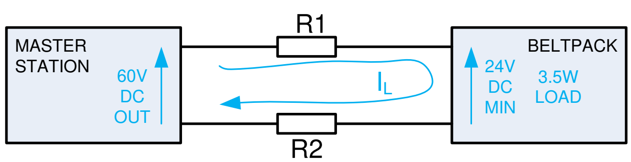


Figure 33: Digital Partyline Voltage Drop

This means that the calculation of maximum cable length is different to that for an analog beltpack. First it is necessary to calculate the current (I_L) that would flow in the circuit with the lowest acceptable voltage (V_L) available at the load:

$$I_L max = \frac{P_L}{V_L}$$

$$I_L max = \frac{3.5}{24} = 146\ mA$$

then calculate the maximum resistance:

$$R1 + R2 = \frac{V_{psu} - V_L(min)}{I_L}$$

$$R1 + R2 = \frac{59-24}{.146} = 240\Omega$$

and maximum distance allowed as before:

$$Dmax = \frac{\mathrm{R_{MAX}}}{\mathrm{R/Km}}$$

$$Dmax = \frac{240}{2 \text{ x } 33} = 3.6Km$$

Again, if the number of units at the end of the line increases from one to **ten beltpacks**, the maximum cable length allowable (with **Belden 9463F**) drops to:

$$Dmax = \frac{V_{psu} - V_L min}{(\frac{P_L}{V_L}) \text{ x R/Km}}$$

$$Dmax = \frac{59-24}{(\frac{35}{24}) \text{ x } 66} = \mathbf{0.36Km}$$

The nature of Clear-Com's digital partyline system means that when it reaches its maximum cable length for power distribution, units towards the end of the line will cease to operate until the cable connecting them to their Main Station is brought back within the required specification.

5.4 **Data Limits**

In addition to delivering power, in both analog and digital party line systems it is also necessary to ensure that the required data signal can be passed from source to destination.

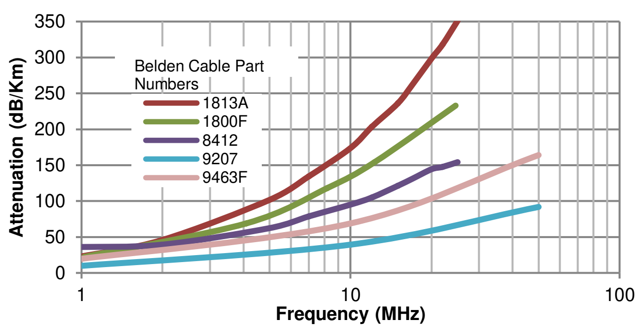


Figure 34: Comparison of Cable Attenuation

As shown in Figure 34, the attenuation of a cable over a given distance increases as transmission frequency increases. Figure 34 also shows us that some cable types are designed to be more suitable for transmitting higher frequency data whilst other cable types are designed to be suitable for lower frequency transmission.

5.4.1 Analog Partyline

With analog party line, the maximum data frequency of interest is the analog audio signal, a frequency of less than 20 kHz. Figure 34 shows us that at frequencies below 1 MHz, all the cable choices shown have a very low attenuation (<30dB/Km).

The nature of the analog partyline system is also such that as cable attenuation between nodes increases, the effect observed by the user is a decrease in audio level heard at the headset with a low distortion audio signal still being received.

Take our previous analog partyline system as an example:

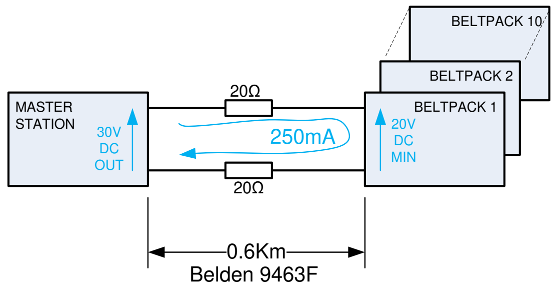


Figure 35: Ten Beltpack Analog Partyline Example

In this extreme case, assume the audio signal is 20 kHz and the cable attenuation at 20 kHz is the same as at 1MHz (In reality it will be much lower, but cable manufacturers do not specify cable attenuation at such low frequencies). There will be attenuation (see Figure 34) of around 30dB/Km so with our maximum allowable cable length the audio level will be reduced by:

$$Attenuation = 0.6Km\, x30 \frac{dB}{Km} = \ 18dB$$

5.4.2 Digital Partyline

In Clear-Com's digital partyline system, higher frequencies are used to transmit digitized audio and control data than are used in analog partyline systems. The maximum frequency used for transmission is around 25MHz.

At that frequency, cable attenuation increases but the nature of the transmission and reception technology used means that the system can operate with much greater levels of signal attenuation without loss of data.

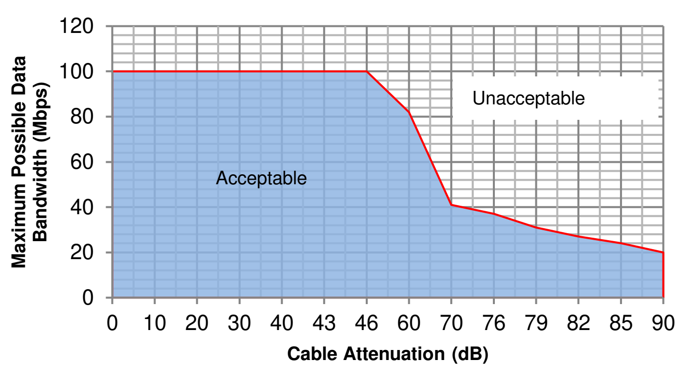


Figure 36: Clear-Com Digital Partyline Maximum Bandwidth vs Attenuation

Figure 36 shows us that depending on the amount of data transmitted, Clear-Com's digital partyline system can operate with as much as 90dB of attenuation between nodes.

It should be noted that this ability to receive highly attenuated signals increases the possibility of crosstalk between cables. It is therefore important to maintain cable shield integrity through all connectors, splitter boxes and patch panels.

Each unit connected to a digital partyline requires a certain amount of bandwidth available on the line in order to operate. Clear-Com's digital partyline products each require the following bandwidth:

Description	Model #	Bandwidth (Mbps)
HelixNet Beltpack	HBP-2X	1.8
HelixNet Beltpack	HBP-2XS	1.8
HelixNet Speaker Station	HKB-2X	1.8
HelixNet Remote Station	HRM-4X	3.1

Table 7: Clear-Com Digital Partyline Unit Bandwidth Requirements

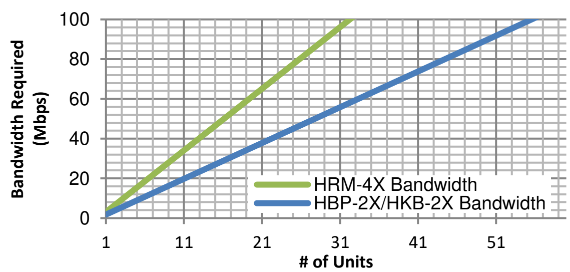


Figure 37: Bandwidth required vs Number of Units

Figure 37 shows the bandwidth required on a Clear-Com digital partyline for differing numbers of digital partyline units.

Take our previous digital partyline example:

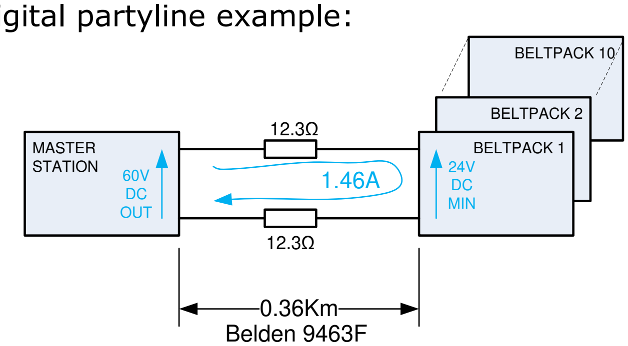


Figure 38: Ten Beltpack Digital Partyline Example

This system, with .36Km of cable with attenuation (from Figure 34) of around 130dB/Km will have an attenuation of:

$$Attenuation = 0.36Km\, x 130 \frac{dB}{Km} = \; 48dB$$

From Table 7 we can calculate that these 10 beltpacks will require:

$$Bandwidth = 1.8Mbps\; x\; 10 = 18Mbps$$

From Figure 36 we can see that with 18Mbps of bandwidth required, the system will operate with attenuation up to 90dB. Our calculated attenuation for this example of 48dB above is well below that limit.

The nature of Clear-Com's digital partyline system means that when it reaches its maximum cable length for data attenuation, the user will begin to experience lost audio packets. When the maximum cable is exceeded by a significant amount, units will fail to connect to their Main Station. The units all provide a signal level meter on their display indicating the quality of connection back to their Main Station helping a system installer to see when they are approaching the system's limits.

5.5 Conclusions

Cable Types

Analog and digital partyline intercom systems place different demands on the cables used and whilst some cables are very good for both systems, some that are fine for one are not good for the other.

- Analog systems require good shielding and little attention needs to be paid to attenuation at higher signal frequencies.
- Digital systems require particular attention be paid to attenuation at frequencies into the MHz range.

Due to the combination of their low DC resistance and low attenuation at 25MHz data rates, Clear-Com recommends the use of:

- *Belden 9207 cable for fixed digital partyline installations.*
- *Belden 9463F cable for portable or temporary digital partyline installations.*

See Belden's website at: www.belden.com for more details.

It is worthy of note that star quad cables have very high attenuation at 25MHz making them particularly poorly suited to digital partyline applications.

Limit Conditions

The nature of analog and digital systems dictates that their failure mechanisms are different when cable requirements aren't met.

Figure 39 and 40 below give a comparison of what a user can expect from analog and digital partyline systems when approaching the limits of their cable's capabilities.

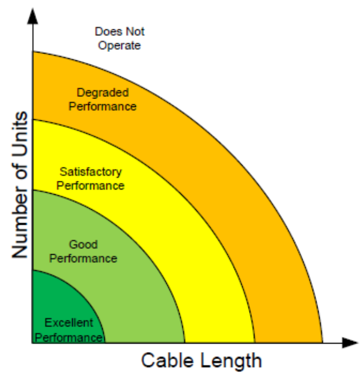


Figure 39: Analog Partyline Performance with increasing Cable Length and Number of Units

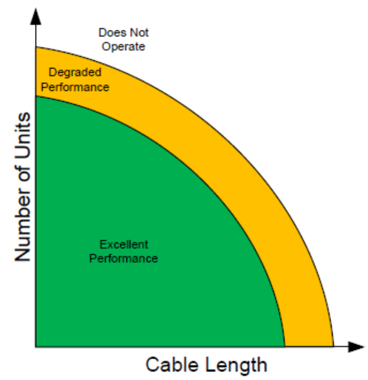


Figure 40: Digital Partyline Performance with Increasing Cable Length and Number of Units

5.6 Notes on HelixNet Powerline and Patch Bays and Mult Boxes

While the concept of HelixNet partyline was to keep the platform familiar and to be used with existing infrastructure there are some considerations to be followed when wiring the powered output (powerline) thru patch bays and mult boxes with daisy chained connectors.

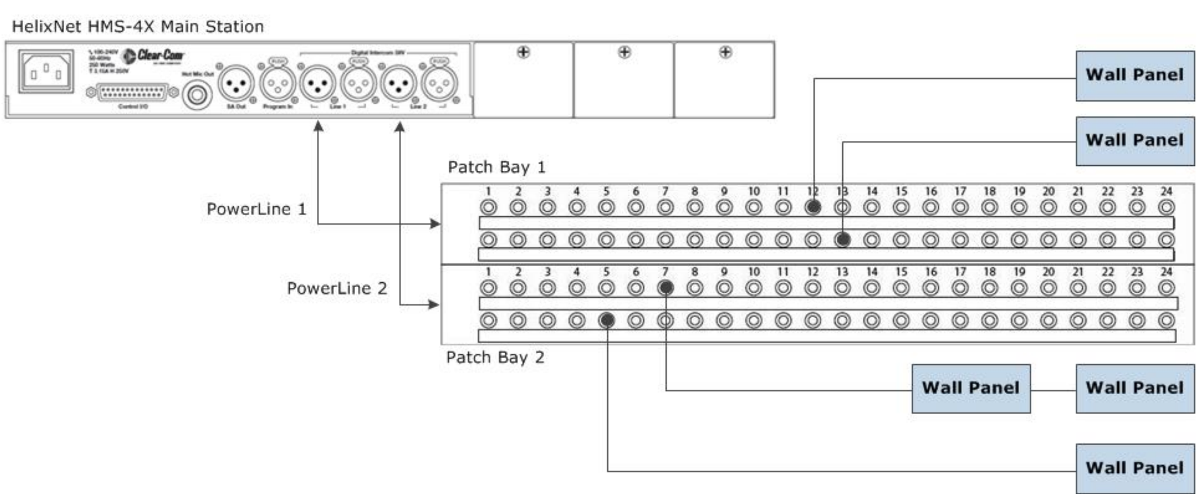


Figure 41: Patch Panels

Clear-Com recommends XLR patch panels. These should be made of 3 pin XLR feed-through adapters (for example, Neutrik NA3MDF) that maintain shield integrity from the back to the front. These adapters also enable easier rewiring of the back or the front of the panel.

Figure 42: XLR Patch Panels

Splitter boxes

Ensure that you split the digital intercom lines within a shielded enclosure. For example, a 1RU shielded chassis, such as Middle Atlantic CH1, and daisy-chained XLR connectors (for example, Neutrik NC3MD or NC3FD).

To prevent crosstalk between lines, ensure that each shielded splitter box contains only one digital intercom line. If you should split more than one digital intercom line use multiple shielded enclosures.

5.7 Recommended Wire chart for all PL

Part #	Pairs	AWG/stranding	O.D. Inch	Single Capacitance pf/ft	.1uf Distance Feet	Shield Resistance Ω per 1000 feet	Fire Rating	Application
9463F	1	20AWG(42x36)	0.243	36	2777	5.00	CM	Portable
1800F	1	24AWG(42x40)	0.221	26	3846	6.42	C12R	Portable
1696A	1	22AWG(7x30)	0.235	26	3846	5.90	NONE	Portable
8412	1	20AWG(26x34)	0.262	58	1724	4.50	NONE	Portable
9207	1	20AWG(7x28)	0.330	23	4347	1.74	CMG	Portable
8428	1	18AWG(41x34)	0.290	60	1667	2.70	NONE	Portable
3074F	1	18AWG(7x26)	0.460	25	4000	2.83	CMG	Portable
8408	1	16AWG(65x34)	0.380	55	1818	2.40	NONE	Portable
1505A	coax	#20 solid	0.233	16.3	6135	3.80	CMG	Fixed
8762	1	20AWG(7x28)	0.204	49	2041	10.50	CM	Fixed
8760	1	18AWG(16x30)	0.222	44	2273	20.50	CM	Fixed
Typical Star, Quad	—	2X24	0.236	62	1612	9.10	—	—
1634A	2	22AWG solid	0.431	17	5882	2.30	CMG	Fixed
6543PA	4	22AWG(7x30)	0.252	45.5	2020	27.00	Plenum	Fixed

Wire Gage	#10	#12	#14	#18	#20	#22	#24	#26
Ω per 1000 feet	1	1.5	2.6	4.2	10.5	16.8	27	42

1 amp through 4.2 ohms = 4.2 volts drop

amps x resistance = voltage drop

Table 8: Recommendations for various wire types

6 GLOSSARY

Some of the terms used when discussing production intercom for television or theatre may be new to you as they are unique to intercom applications. Although many of the terms are common to other audio applications, to be certain you understand their meanings we offer the following definitions:

A1: Person responsible and in charge of the audio effort. Usually the A1 is the person who mixes the show.

A2: This person works for the A1 and is generally responsible for all audio setup outside of the venue or production truck

Active: As opposed to **passive**, this means that electronic or electric circuitry is involved in accomplishing something.

A/D: Analog-to-digital conversion.
ADC: Analog-to-digital converter

All Call: Activating an All Call button from a Main Station or Remote Main Station or user station will initiate a talk to all channels at once.

Ambient Noise: Those background sounds which are not part of the specific communication but are picked up by the microphone. Selection of a good noise-canceling microphone will reduce ambient noise.

Ampere: The amount of electrical current when one volt is applied to one ohm.

Amplifier: Usually an electronic device that increases the amplitude of an electrical signal. Examples include a microphone preamplifier that brings millivolt signals to volt levels.

Audio Frequency: Range of frequencies lying within the range of human hearing, often 20 hertz to 20,000 hertz, where hertz is cycles per second.

AWG American Wire Gauge: a standardized wire gauge system used for measuring the diameters of round, solid, nonferrous, electrically conducting wire. The AWG of a stranded wire is determined by the total cross-sectional area of the conductor, which determines its current-carrying capacity and electrical resistance. Increasing gauge numbers give decreasing wire diameters.

Balanced Audio: Audio that is differently driven down a path, where neither lead of the audio pair is at ground potential. Each signal on the two leads is 180 degrees out of phase with each other. Because of this fact balanced audio is generally immune to outside interference. Audio XLR connectors have three conductors, two for each of the two audio signals and one for ground. Compared to **single ended** audio where the audio signal is on a single conductor and referenced to ground, which is the other conductor. **RCA** audio connectors carry single ended audio.

Balance Line: Defined in terms of the impedances of the two signal conductors with respect to a reference, which is usually "ground". Neither conductor is tied to circuit common. Circuit common is either tied to a transformer center tap, or is an electrical center point, or not tied at all.

Beltpack: A portable headset user station. This station is designed to be worn on a user's belt with the idea of semi-portability. It can be single or two intercom channels capable. Requires a headset or handset. Interconnects to system with microphone cable and is powered by a central Power Supply or Main Station.

Breakout Box: A box with multiple connectors on it that is connected to a cable that encompasses multiple feeds within it. The breakout box provides a separate connector for each of the individual feeds for the signals in the cable. See mults and pigtails.

Biscuit: A portable intercom user speaker station.

Bridging, High Impedance (hi-Z): A method of connecting to a nominal impedance audio line (such as Clear-Com) and creating a non-significant effect on the circuit. (loading or taking appreciable power from that line.) Simply stated, as you add more and more stations to the line, the volume remains constant.

Binaural: Refers to two ears. In intercom use it refers to two signals, may be stereo or may be two different signals fed to two ears independently. Sometimes called split-ear.

Bus: Circuitry that transports multiple digital signals grouped together as parallel lines. Also refers to large conductors used to carry electric power. Such a wire, or in general, a collection of wires that carry some multi-bit information, is called a bus.

Call Signaling: This feature is included with the majority of Clear-Com products. It can be an audible and/or visual alert on a user station (a lamp or LED) used to attract the attention of an operator signifying that someone at another station wants to initiate a conversation. The Call light feature is used for two different purposes: 1) primarily to get a user's attention of incoming communication. 2) To indicate a cue, routinely: light on means standby, light off after light on means apply.

Cans: Slang for headphones.

Capacitance: Exists between the transmission line wires. The capacitance between wires is usually expressed in Pico farads per unit length. This electric field between the wires is similar to the field that exists between the two plates of a capacitor. Your cable consists of two wires normally twisted together in a bundle. Between any two wires there will be capacitance. High capacitance will affect the frequency performance of a line in two ways - increased attenuation and changing phase of a signal.

Channel: A intercom "channel" is one individual circuit of communication used within a partyline - it is typically a two-way talk/listen path. This applies to wired and wireless partyline systems. An example would be a partyline channel for spot light operators. There can be more than one line circuit channel to allow for multiple conversations or information flows to occur simultaneously. It is possible for a user station (beltpack) to select between several channels available in a system with a channel selector on the user station. This allows for multiple conversations or information flows to occur independently as needed. An example would be a Remote Stage Manager with Carpenters on Channel A and the rest of the production on Channel B.
**Channel is also used to describe a range of frequencies (or, equivalently, wavelengths) assigned by a government for the operation of a particular television station or radio station. In common usage, the term also may be used to refer to the station operating on a particular frequency. This is common with two-way radios.*

Circuit: A complete path for electrical power or an electrical signal (usually two conductors). In an intercom system, a channel for one or two-way conversation may be called a circuit.

CODEC: An acronym of Compression, Decompression = a device or piece of software which takes one file or signal format and translates it to another with an ideally undetectable loss of quality. Equipment that takes baseband video and audio and compresses it into one of many file or transport stream formats or decompresses it back to baseband.

Conference Line Intercom: see Partyline

Crossover Cables:

Computers: A crossover cable is sometimes known as a null modem.
A crossover cable is a type of twisted pair copper wire cable for LANs (local area network) in which the wires on the cable are crossed over so that the receive signal pins on the RJ-45 connector on one end are connected to the transmit signal pins on the RJ-45 connector on the other end. This is the opposite of the usual straight-through LAN cable, in which the receive and transmit signal pins on one connector are connected to the corresponding pints on the other connector. Its purpose is to allow the direct connection of two LAN devices, such as two hubs, two switches or a hub and a switch. It can also be used to create a direct connection between two computers.

Audio:
Very similar to a crossover cable used in Ethernet networks an audio crossover cable is a type of twisted pair copper wire cable in which the wires on the cable are crossed over so that the receive audio signal pins on one end are connected to the transmit audio signal pins on the other end connector.

Cross Talk: Leakage of audio transmissions from one channel to another.

Decibel (dB): In electronics and communications, the decibel is a logarithmic expression of the ratio between two signal power, voltage, or current levels. In acoustics, the decibel is used as an absolute indicator of sound power per unit area.

Dry Pair: A telephone term is used to describe a pair of wires (2 conductors) that carry audio but no voltage. Contrast this with a "Wet Pair" that carries both audio and voltage.

Daisy Chain: Some Clear-Com Partyline user stations allow the looping together (or daisy chaining) of user stations. These stations have a "loop-thru" connector as well as a 'line" or "line input" connector. A partyline system can be constructed by connecting one user station to another via the line and loop-thru connectors. Other wiring options are "home running," which is running a line cable from each user station to a central point connecting to the power supply ("home").

Destination: A destination is anything that a talk key talks to or a listen key listens to. Therefore, a destination can be an intercom station, beltpack or interface, (or group of such devices connected together), which is assigned to a source channel of a power supply or central intercom Main Station. Within foldback systems for cuing talent, a destination also describes the talent receiver or loudspeakers within green rooms

Dim: This the intentional attenuation of an audio signal. "Dim" occurs in two contexts in intercom systems. First, Dim is used to correct a feedback problem that can occur between two user speaker panels operating in close proximity that talk/listen to a common destination. This can help to prevent occasional feedback between the speaker and microphone due to volume settings, microphone placement, etc. Second, dim is referred to the lowering of a program feed to a destination so that a talk path could be heard – such as in a dressing room page or talent IFB cueing.

Dual Listen: This is an option or feature of intercom user stations. Dual listen permits an operator to listen to two channels at once. This may be a mix of two channels to one ear, or in a binaural or stereo user station, one channel can be assigned to one ear and the other channel to the other ear.
Dual Listen could also be an intercom channel and a program audio source. The dual listen pots are functionally configured in two ways: 1. One pot controls the audio of the channel actively used, and the second pot controls the audio of a monitored channel. 2. One pot is always one channel and the other pot is always the other channel.

Duplex: Duplex refers to bi-directional communications. **"Full" Duplex** describes bi-directional communications all the time. Regular communications between individuals conversing face to face is "full duplex" -- in other words you can talk and listen simultaneously. Full Duplex communication allows simultaneous two-way conversations, plainly - one person can interrupt the other.
The alternative is **"Half" Duplex.** Half Duplex communication allows two-way conversations, one-way at a time, such that one person cannot interrupt the other. A walkie-talkie is a good example of half-duplex communication.

EFP: Electronic Field Production. An EFP truck contains the necessary audio, video, intercom, and other equipment to create these productions.

EMI: Electromagnetic Interference. Interference caused by the radiation of electrical or magnetic fields
from sources such as radio transmitters, light dimmers, computers, and transformers.

ENG: Electronic News Gathering. An ENG truck contains the necessary audio, video, intercom, communications, and other equipment to effectively support gathering news and transmitting news reports back to a studio.

Four-Wire: A communications system where the paths are different for talk and listen. In intercom channels there are four wires (two paths). Four-wire systems can be four-wire balanced and four-wire unbalanced. Four-wire audio is more or less defined as a pair of conductors carrying an input/receive signal and a second pair carry the output/send signal. The four-wire circuit gets its name from the fact that a balanced pair of conductors was used in each of two directions for full-duplex operation.

> **Four-Wire Unbalanced:** A four-wire system that uses a circuit common and two additional conductors. The talk pathway consists of one conductor plus circuit common. The listen pathway consists of another conductor and circuit common.
> **Four-Wire Balanced:** Four-wire balanced is similar to four-wire unbalanced except that conductors are not tied to circuit common. Circuit common is either tied to a transformer center tap, or is an electrical center point, or not tied at all.

GHz – Gigahertz: Thousand million cycles per second.

GPIO: General Purpose Input/Output. (You may also see this referred to simply as "GPI".) GPIO is a means of controlling devices using switch contact closures, DC voltages, or similar methods. Additional general purpose control outputs are provided by optional relay frames.

Headset: portable intercom connection from a user station to one or both ears via headphones with integrated microphone on a boom arm. Connects to a beltpack, remote stations or Main Stations. Used by the user to talk and or listen.

Headset Microphone Type:

> **Dynamic Microphone:** Converts sound pressure waves to electrical signals by means of a coil attached to a diaphragm moving in a magnetic field.
> **Electret Microphone:** A microphone using a capacitor as the sound pressure sensing element. Electret microphones are a special case of condenser microphones in that they are permanently polarized and require no special polarizing voltage. Electret microphone outputs are high impedance.

Headset/Double-Muff: Headset with two earphones plus a microphone. It can be connected monaurally (same information, both ears) or binaurally (separate feed each ear). In binaural operation, the feed can be intercom in one ear and program in the other, or intercom channel A in one ear and intercom channel B in the other ear. Channels A and B are either conference line channels or other intercom feeds. A binaural feed requires a binaural/stereo capable user station.

Hertz (symbol: Hz): a unit of frequency. It is defined as the number of cycles per second. It is the basic unit of frequency in the International System of Units (SI), and is used worldwide in both general-purpose and scientific contexts. Hertz can be used to measure any periodic event; the most common uses for hertz are to describe radio and audio frequencies, more or less sinusoidal contexts in which case a frequency of 1 Hz is equal to one cycle per second.

IFB: is abbreviation for Interrupted Fold-Back. It is often a communication circuit feature or a separate system that interfaces with the intercom system. In use, a user, (typically talent) listens to the program all the time and is 'interrupted' by the director (typically) with cues or instruction.

Impedance: Impedance is the total effect of resistance, capacitance and inductance and each of these oppose electrical flow on a cable in different ways. Impedance is a combined effect or total opposition to current flow.

I/O: input and output connections

ISO: A private conversation path. An ISO channel allows one to simply push a button and transfer themselves and the person they wish to speak with to an isolated channel.

kHz – Kilohertz: Thousand cycles per second.

Local area network (LAN): a computer network covering a small physical area, like a home, office, or small group of buildings, such as a school, or an airport. The defining characteristics of LANs, in contrast to wide-area networks (WANs), include their usually higher data-transfer rates, smaller geographic place, and lack of a need for leased telecommunication lines.

Latency: Usually used in context of audio, video, and RF signals. Latency is the time it takes a signal to travel over a path or through a piece of equipment.

Limiter: Used to limit dynamic range to ensure adequate intelligibility to the listener. The limiter/compressor in user stations has three functions: I) It helps loud talkers and soft talkers to be heard equally well, 2) It prevents a loud voice from being severely distorted, 3) It keeps the voltage levels from exceeding system limits.

Line: A single communication or audio path.

Local Power Option: Local Power Source is a small AC converter that converts AC line power to low voltage in order to power a user station --a separate connector is provided. User stations usually get DC from the converter, although occasionally low voltage AC power is used.

Loop-Through See "Daisy Chain".
.
MHz: Million cycles per second.

Main Station/Master Station: Typically a analog or digital partyline multi-channel intercom station with an internal power supply which can provide power both to itself and to all of the remote stations connected to it.

Mix-Minus Bus /feed: A mix-minus feed is typically used for the IFB.
The mix-minus allows the talent to hear the program audio that includes the voices of other talents at other venues as example, but not the talent's own voice. The effect is to allow more normal conversations, on air, among the performers. The bus feed refers to the mixer mix-minus feed available to one or more IFB program inputs.

Mono: Single channel audio

Mults = A large cable that is made up of many smaller internal cables. Often used to carry multiple audio feeds. In many instances fiber optic cables are replacing these cables. A mult box is sometimes called a 'press box' usually reserved for the sports media's section of an arena or frequently at press conferences.

Multi-Channel: More than one channel

Null: A hybrid's ability to isolate the transmit signal from the receive signal in the 2-wire-to-4-wire interface is critical. The quality of this isolation is technically known as return-loss.
A side tone **nulling** control fine tunes the circuitry to best match the devices to the acoustic conditions near the intercom, as well as to the electronic conditions on the intercom line.
They should be set at the time of system installation and adjusted as is comfortable for the user.
Four-wire audio is more or less defined as a pair of conductors carrying an input/receive signal and a second pair carry the output/send signal. This hybrid circuit connects the four-wire audio to the single wire in such a way as to variably restrict the user's reception of his own voice on the intercom line, which is often referred to as "side tone".
High gain between the send and receive poses a risk of oscillation or 'howling' resulting from acoustic and/or electronic coupling within a headset or between a speaker and a microphone.
With manual nulling devices there are the following accessible controls: Separate "R" (Resistance), "L" (Inductance), and "C" (Capacitance) controls compensate for each component of the line's impedance, providing the best null possible.
Ideally, there should be no portion of the talk signal in the listen signal.
The variation of the 2-wire line phase coherency is affected by the cable capacitance (length of wire and gauge) and inductive elements of the line .
The "C" control compensates for cable capacitance; the setting depends on the length of the line.

The “L” control compensates for the low-frequency inductive and capacitive elements the wiring of the external party line presents to the line.
The importance of 2-wire termination, lack of, or double termination also influences the null result. If the “R” control is turned fully counter-clockwise, the line has either more than one termination, or an excessive resistive load. If the “R” control is fully clockwise, then the line has no termination.

Party Line (PL): Intercom system where all people talking on the system can talk or listen to each other simultaneously. The name **PL** (**partyline)** came from the original telephone systems where everyone shared the same line and could hear and join all conversations at once. It is often called **conference systems**, **2-wire** or **TW**, which stands for two-wire (see 2-wire description). A Partyline system allows a group of people to intercommunicate. A Party Line is classically used when several users, such as beltpacks, are active in a common task and they must communicate with each other all the time.
Characteristics of party lines

- When the party line is already in use, if any of the other subscribers to that line pick up the headset, they can hear and participate in the conversation.
- Completely non-private lines
- Systems are created from building block components to correspond the demand of the event.

Conference systems can be distributed or centralized. Most of the systems are distributed conference systems. Distributed means that a station can be plugged-in at any arbitrary point along the bus or channel.

Pigtail: A group of cables that converge into one large connector at one end and at the other ends each cable has its own separate connector. It is designed to patch into an existing line or to terminate the ends of a long run
Point to Point: One path to one person

Powerline Technology: A method where data is modulated on a carrier frequency and added to electrical wiring to simultaneously carry both data and electric power. It is a highly-effective data transmission medium.

Power Supply: The source of electrical power (“power outlet”). In North America this source is generally 120 volts AC, 60 hertz. In Japan the source is generally 100 volts, 50 or 60 hertz. In Britain the source is 240 volts, 50 hertz. In Europe the power is usually 220 volts, 50 hertz. In addition, some equipment is operable off of DC sources such as batteries.

Power Supply Clear-Com: A specific power supply to operate Clear-Com beltpacks and remote stations. The analog Clear-com power supply provides low noise DC power (30 VDC ± 0.5V) up to 1.2 amperes per channel and an audio line impedance of 200 ohms.

Program: Audio source that is fed into the intercom channels or stage announce.

Program Interrupt: Disconnects the audio source while the talk button on the Main Station is pushed. (see IFB)

Push- To- Talk (PTT): Usually used on handsets or push-to-talk microphones. Pushing the button enables the microphone talk circuit on.

QOS: Quality Of Service

Rack Unit: A unit of measure used in racks that designate how tall a piece of equipment is when installed in a rack. One rack unit or RU is 1.75 inches tall (44.45 mm). A piece of equipment that is said to be 4 rack units tall has a height of 7 inches.

Relay: A relay is an electrically operated switch. Commonly these relays are Normally-open (**NO**) contacts which mean they connect the circuit when the relay is activated; the circuit is disconnected when the relay is inactive. It is also called a **Form A** contact or "make" contact. Typically, in audio electronics, these relays are of the dry contact type. Dry contact refers to a contact of a relay which does not make or break a current. They simply turn something on or off.
Rating: In audio electronics a rating of .05 to 2 Amp at 24 volts AC/DC maximum is common. A "Phoenix" type connector plug is also common and it plugs into the relay contact port on the rear of the base station for wiring to external devices. A use for a relay is sometimes associated with turning on a light for attention such as an on-air light.

Remote Mic Kill (RMK): The ability for certain intercom user stations to shut off all talk circuits on an intercom line in a system.

Remote Station: Like the beltpack, this would be any of the products connected to the intercom line that allow duplex or half-duplex conversation, but do not contain a Power Supply. A Remote Station cannot power other Stations

RF: Radio Frequency.

Sidetone: This is a small amount of your own voice heard in your earphone as you are speaking.

Splitter: Usually refers to a **passive** audio device that takes one audio source in and provides two or more outputs.

Stage Announce (SA): Typically a voice page made over a loudspeaker. In wired intercom when a SA control is pressed, either at a base station or an assignable beltpack, the user's audio is routed to the stage announce connector on the back of the base station. This is usually an analog line level audio output. The user may lose their headset side tone as an indication that stage announce is activated. The other users do not hear the announce audio. The button is non-latching.

Station: A station is connected to one or more channels. For example, if you have six people who need to hear one director you have a seven-station single-channel need. If the same director needs to speak privately to any one of the six, add a second channel. You now have a seven-station, two channel system.

Termination: Passive network that is connected in each channel, usually on the Power Supply or Main Station.

Tie Lines = Generally cables that have no dedicated use that link one section to another. Tie lines usually terminate at patch panels or other IO panels.

Two-Wire: A communications system where the path is the same for both talk and listen. In intercom channels there are two wires (one path). Two-wire systems can be two- wire balanced or two-wire unbalance

Turn around: A term used to describe an audio interface cable or barrel type tube device, sometimes called couplers, that coverts a female-to-female or male-to-male connector used to turn snake channels from a send to a return or vice versa, but they also come in handy when a stage hand has inadvertently run a very long XLR cable in the wrong direction.

Wet Line: An intercom that carries both audio and DC voltage/current. As opposed to a dry line that carries only the audio.

XLR: standard connector for audio, most commonly with three pins or contact points. Four and five pin versions are used for intercom headsets and multi-channel microphones. Cannon Electric invented the connector and made it part of its "X" series or connectors. Because of its locking mechanism the "L" was added, and the fact that it has rubber insulation resulted in the final "R" designation.

7 KEY NETWORKING TERMS

Internet Protocol (IP) - is responsible for addressing hosts and for routing datagrams (packets) from a source host to a destination host across one or more IP networks. For this purpose, the Internet Protocol defines the format of packets and provides an addressing system that has two functions: Identifying hosts and providing a logical location service.

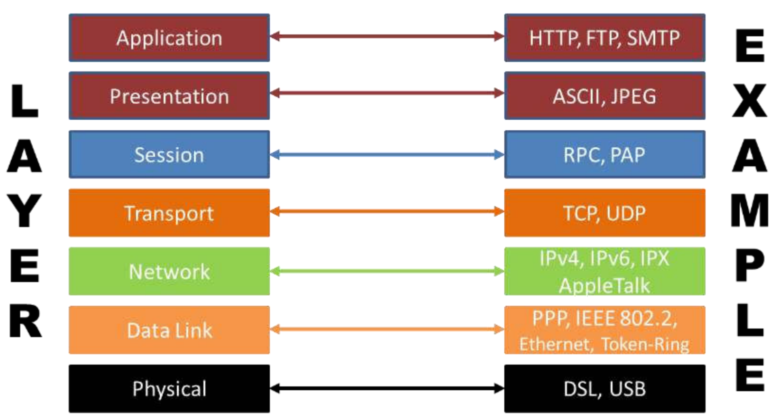
Transmission Control Protocol (TCP) - provides a communication service at an intermediate level between an application program and the Internet Protocol. It provides host-to-host connectivity at the Transport Layer of the Internet model.

User Datagram Protocol (UDP) - uses a simple connectionless transmission model with a minimum of protocol mechanism. UDP provides checksums for data integrity, and port numbers for addressing different functions at the source and destination of the datagram. It has no handshaking dialogues, and thus exposes the user's program to any unreliability of the underlying network: there is no guarantee of delivery, ordering, or duplicate protection.

broadcast data (bc) - Directed broadcast packets NOT global broadcast packets which are 255.255.255.255

OSI 7-Layers - Reference Model is derived from ARPANET and initially financed by the Defense Advanced Research Projects Agency, or DARPA, of the US Department of Defense.

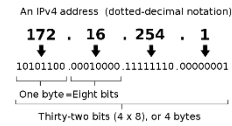
Note: The OSI 7-Layer Model attempts to define how telecom and data systems should interconnect.



Dynamic Host Configuration Protocol (DCHP) - dynamically distributes network configuration parameters, such as IP addresses, for interfaces and services.

Internet Protocol version 4 (IPv4) - uses 32-bit addresses which limits the address space to 4294967296 (232) addresses.

Subnet is a logical subdivision of an IP network. The practice of dividing a network into two or more networks is called subnetting.



–Example - 255.255.255.000

Eg – 192.168.050.025
Mask - 255.255.255.000
192.168.050 < Network
.025 < Device

–Example - 255.255.000.000

Eg – 192.168.050.025
Mask - 255.255.000.000
192.168 < Network
.050.025 < Device

Default Gateway - in computer networking is the node that is assumed to know how to forward packets on to other networks. Typically, in a TCP/IP network, nodes such as servers, workstations and network devices each have a defined default route setting, (pointing to the default gateway), defining where to send packets for IP addresses for which they can determine no specific route.

Domain Name Servers (DNS) are the Internet's equivalent of a phone book. They maintain a directory of domain names and translate them to Internet Protocol (IP) addresses.

Example: You enter www.ClearCom.com the DNS translates IP address 5.79.47.16

Ports - is an endpoint of communication in an operating system. A port is always associated with an IP address of a host and the protocol type of the communication, and thus completes the destination or origination network address of a communication session. A port is identified for each address and protocol by a 16-bit number, commonly known as the port number.

For example, an address may be "protocol: TCP, IP address: 1.2.3.4, port number: 80", which may be written 1.2.3.4:80 when the protocol is known from context.

Note: Port numbers are in the range 0 – 65536. Web servers are by default Port 80 while FTP servers are typically Port 21.

Port forwarding - is an application of network address translation (NAT) that redirects a communication request from one address and port number combination to another while the packets are traversing a network gateway, such as a router or firewall. This technique is most commonly used to make services on a host residing on a protected or masqueraded (internal) network available to hosts on the opposite side of the gateway (external network), by remapping the destination IP address and port number of the communication to an internal host.

Virtual private network (VPN) - extends a private network across a public network, such as the Internet. It enables users to send and receive data across shared or public networks as if their computing devices were directly connected to the private network. Applications running across the VPN may therefore benefit from the functionality, security, and management of the private network.

Types:

Internet Protocol Security or IPSec - is used to secure Internet communication across an IP network. IPSec secures Internet Protocol communication by authenticating the session and encrypts each data packet during the connection.

Layer 2 Tunneling Protocol (L2TP) - or Layer 2 Tunneling Protocol is a tunneling protocol that is usually combined with another VPN security protocol like IPSec to create a highly secure VPN connection.

Point – to – Point Tunneling Protocol (PPTP) - creates a tunnel and encapsulates the data packet. It uses a Point-to-Point Protocol (PPP) to encrypt the data between the connection. PPTP is one of the most widely used VPN protocol and has been in use since the time of Windows 95. Apart from Windows, PPTP is also supported on Mac and Linux.

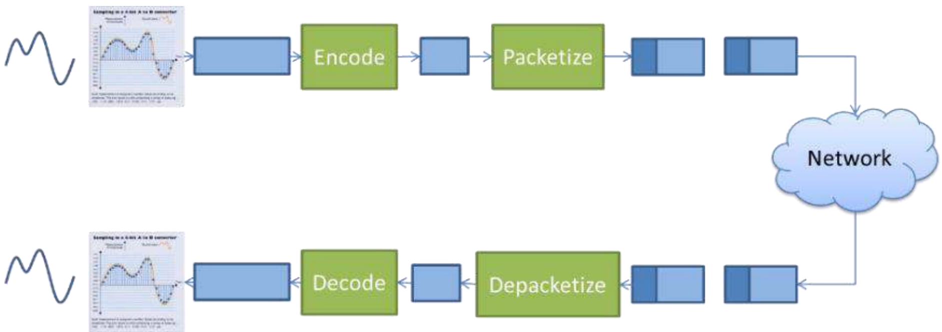
Secure Sockets Layer (SSL) and **Transport Layer Security** (TLS) - create a VPN connection where the web browser acts as the client and user access is restricted to specific applications instead of entire network. SSL and TLS protocol is most commonly used by online shopping websites and service providers. Web browsers switch to SSL with ease and with almost no action required from the user, since web browsers come integrated with SSL and TLS. SSL connections have https in the beginning of the URL instead of http.

OpenVPN - is an open source VPN that is useful for creating Point-to-Point and Site-to-Site connections. It uses a custom security protocol based on SSL and TLS protocol.

Secure Shell (SSH) - creates the VPN tunnel through which the data transfer happens and also ensures that the tunnel is encrypted. SSH connections are created by a SSH client and data is transferred from a local port on to the remote server through the encrypted tunnel.

Quality of Service (QoS) - is the overall performance of the computer network, particularly the performance seen by the users of the network.

Codec - is a device or computer program for encoding or decoding a digital data stream or signal.



Bandwidth - is the bit-rate of available or consumed information capacity expressed typically in metric multiples of bits per second.

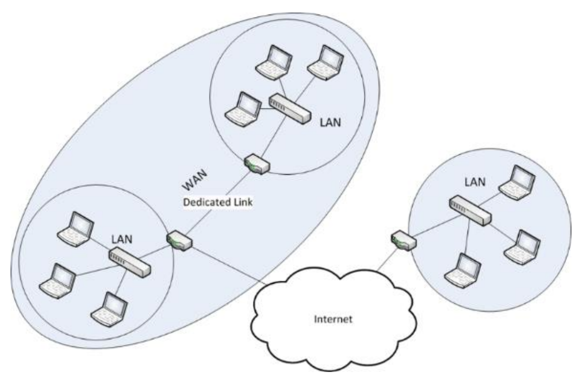
Jitter - is a common problem of the connectionless networks or packet switched networks. Because the information (voice packets) is divided into packets, each packet can travel by a different path from the sender to the receiver. When packets arrive at their intended destination in a different order then they were originally sent, the result is a call with poor or scrambled audio.

Dante - Based on industry standards, Audinate created Dante, an uncompressed, multi-channel digital media networking technology, with near-zero latency and synchronization.

For more training and information on Dante visit www.audinate.com/resources/training-and-tutorials

Local Area Network (LAN) - everything on the same segment using local sets of IP switches to local PCs and equipment.

Wide Area Network (WAN) - may contain several LANs connected over public networks (e.g. T1) or between sites through managed routers. The WAN may have a selection of QoS depending on priority, type of service and packet characteristics.



Internet - is an internationally connected network of billions of smaller WANs connecting businesses and homes.

Multicast - is group communication where information is addressed to a group of destination computers simultaneously. Group communication may either be application layer multicast or network assisted multicast, where the latter makes it possible for the source to efficiently send to the group in a single transmission. Copies are automatically created in other network elements, such as routers, switches and cellular network base stations, but only to network segments that currently contain members of the group.

Internet Group Management Protocol (IGMP)/Snooping - is a communications protocol used by hosts and adjacent routers on IPv4 networks to establish multicast group memberships. IGMP is an integral part of IP multicast. IGMP can be used for one-to-many networking applications such as online streaming video and gaming, and allows more efficient use of resources when supporting these types of applications.

Broadcast - is the distribution of audio and/or video content or other messages to a dispersed audience via any electronic mass communications medium, but typically one using the electromagnetic spectrum (radio waves), in a one-to-many model.

Session Initiation Protocol (SIP) – is a communications protocol for signaling, for controlling multimedia communication sessions. Internet telephony, business IP telephone systems, service providers and all of the carriers use SIP. SIP can be used to set up and control voice and video calls, as well as instant messaging. The most common application of SIP is the setup and termination of Voice over IP (VoIP) telephone calls.

Voice over IP (VoIP) - is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.

Real-time Transport Protocol (RTP) - is a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features. RTP typically runs over UDP.

Auto-negotiation - is an Ethernet procedure by which two connected devices choose common transmission parameters, such as speed, duplex mode, and flow control.

Full-Duplex - both parties can communicate with each other simultaneously.

Network Interface Controller (NIC) - also known as a network interface card, network adapter, LAN adapter or physical network interface. A 'NIC' Is a computer hardware component that connects a computer to a computer network.

Backplane speed (aka: switching fabric or switching capacity) - Is the maximum number of packets/sec that can be routed out of any one port per second. When the forwarding capabilities of a backplane are greater than the sum of speeds of all ports (counted twice, Tx/Rx) we call the switching fabric non-blocking, meaning traffic between a pair of ports is not influenced by what traffic is exchanged on all other ports.

Forwarding Rate - The forwarding rate is expressed in packet per seconds and expresses how many packets per second are needed to reach a certain traffic volume (throughput). As forwarding rate depends on frame size, a switch is normally non-blocking up to a certain frame size.

Throughput - specifically relates to how much data can cross the switch in a given time frame.

SFP – The (Small Form-factor Pluggable) transceiver, also known as **mini-GBIC**, offers a standard optical modular, hot swappable electrical interface, one gigabit port that can support a wide range of physical media, from copper to long-wave single-mode optical fiber, at lengths of hundreds of kilometers.

ABOUT CLEAR-COM

Clear-Com, an HME company, is a trusted global provider of professional real-time communications solutions and services since 1968. We innovate market proven technologies that link people together through wired and wireless systems.

Clear-Com was first to market portable wired and wireless intercom systems for live performances. Since then, our history of technological advancements and innovations has delivered significant improvements to the way people collaborate in professional settings where real-time communication matters. For the markets we serve—broadcast, live performance, live events, sports, military, aerospace and government—our communication products have consistently met the demands for high quality audio, reliability, scalability and low latency, while addressing communication requirements of varying size and complexity. Our reputation in the industry is not only based on our product achievements, but also on our consistent level of customer engagement and dedication to delivering the right solutions for specialized applications, with the expertise to make it work. Around the globe and across markets, Clear-Com's innovations and solutions have received numerous awards and recognitions for ingenuity and impact to customers.

Americas and Asia-Pacific Headquarters
California, United States
Tel: +1 510 337 6600
Email: SalesSupportUS@clearcom.com
SalesSupportAPAC@clearcom.com

Europe, Middle East, and Africa Headquarters
Cambridge, United Kingdom
Tel: +44 1223 815000
Email: SalesSupportEMEA@clearcom.com

China Representative Office
Beijing, P.R.China
Sales/Marketing Tel: +86 10 59002608
Service Tel: +86 10 59000198
Email: SalesSupportAPAC@clearcom.com

www.clearcom.com

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