AN HME COMPANY

Intercom Design Guide

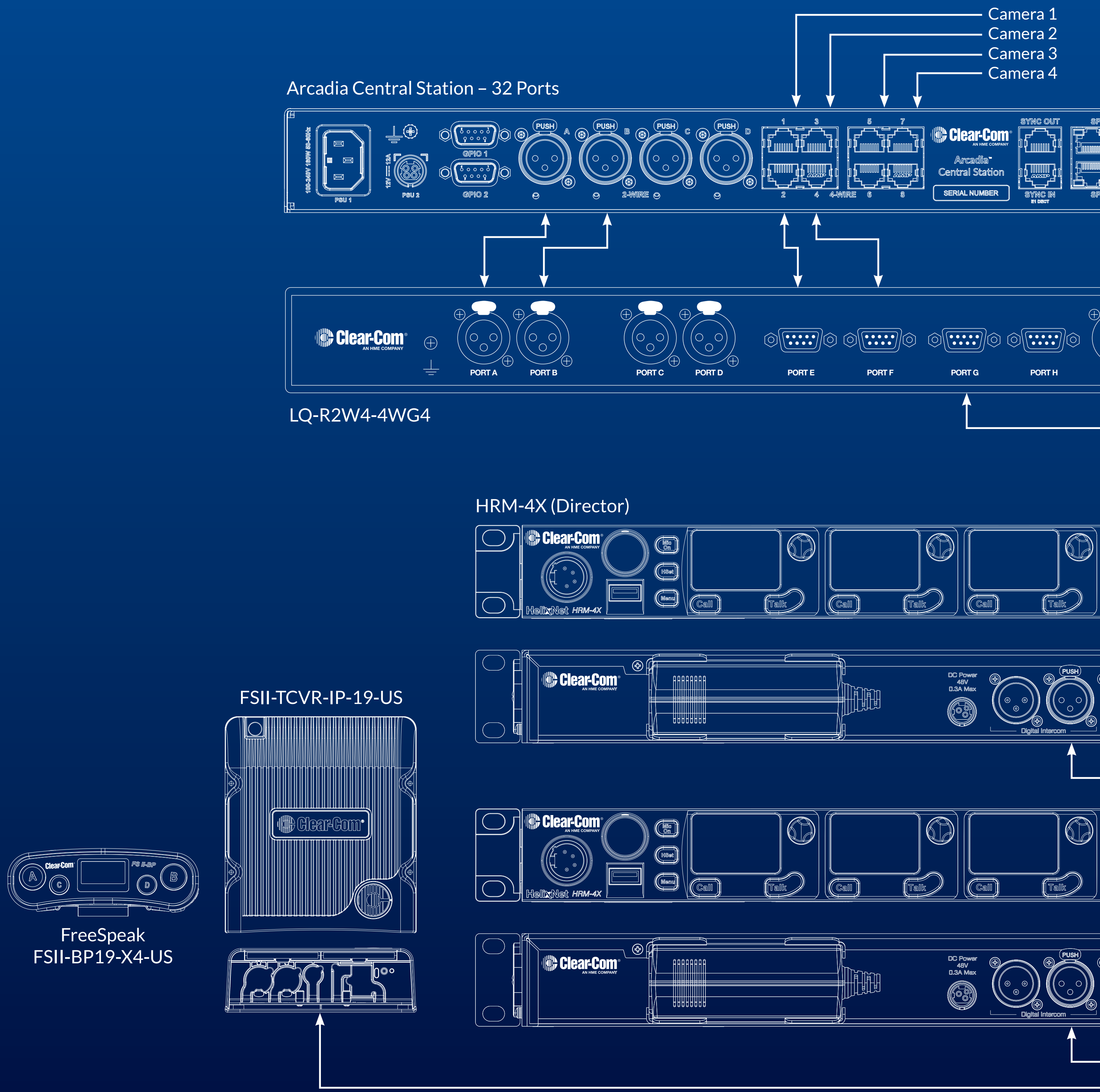


TABLE OF CONTENTS

Thank you for your interest in this paper. This design guide is meant for the person who either seldom spends much time with intercoms or is new to intercoms and needs a starting point to put a system design together.

This document will guide the reader towards conceiving, planning and ultimately designing an intercom system. It includes our best advice and approach on how to compile and record information about a facility, the users, and their workflows.

It would be nearly impossible to try to account for every application where our products can be used, so we will try to describe as many of the most common customer requirements we encounter. We find there is nothing more useful when planning a project than looking at examples to quickly spark thinking. To that end, there are dozens of intercom system diagram examples on the Clear-Com® website listed by applications. You may find a useful template for your facility's ultimate needs by visiting our website's listing of Application Diagrams.

Another useful resource we provide on our website is called the "Solution Finder". This is, in essence as the title describes, a search engine providing 'solutions' to your questions about an intercom topic. You can access the Solution Finder here.

Lastly, for supplemental reading, Clear-Com offers an in-depth guidebook explaining wired intercom technology in general, the theory and appliance as well as most common terms. This e-book is available as a downloadable PDF.

Intercoms are used in many different facilities across a wide range of markets - from Industrial, Healthcare and Education to Performing Arts complexes to television studios for various applications, yet they all undergo a common design methodology based on very simple workflow criteria that we will define. The design of an intercom system, like the design of any key system, must begin with a definition of the individual system's requirements. This definition must include operational, environmental, and budgetary considerations. Most critical of these is operational – i.e., 'what does the user want to do?'

> Within all environments, a full understanding of the user's communication needs is fundamental to designing a satisfactory intercom experience.

We have divided the design guide into four sections. In Section One, we will review the various and most common communication techniques that are built into most modern intercoms. You will find it useful to have this information as you draft your users' workflows and facility layout.

Section Two presents a general overview of the product families which can be combined to create your system. Within this section is important information about IP standards and practices that may require additional reading. We have listed additional material at the end of the section. You should gather from this section how these various products fill the work technique described in Section One and will bring this info as you start your design in Section Three.

In Section Three we will outline steps towards defining the system users and how they work in the facility. Depending on your process, it may be best to reverse that order - define the workplace and people within. Both examples are discussed. Subtle differences but could be the means to a better understanding.

This may be a good point to highlight how understanding user workflows add to understanding communication needs. Across industries we are all familiar with how business workflows are useful for ensuring that important processes are done the right way every time. Within a structured production, show, event or even a meeting we follow either a script or agenda and each player follows a certain workflow to complete their process. Take into consideration how users interact in a facility. As an example, we may find that the spot operators in a show take direction from the lighting director therefore, we can plan that there is a need for a communication channel between them. We also know that the spot operators may need to communicate amongst themselves sometimes called a private channel. Understanding workflows certainly adds to building a solid intercom design.

And in Section Four we present a few sample intercom system designs based on input from users, following the process outlined in the previous sections.

Foreword

Setting aside questions of audio formats and/or transmission platforms, there are several types of production intercom systems and subsystems, including: wired analog partyline systems, wired network partyline systems, wireless partyline systems, matrix systems, interfaces, virtual communications, and communication sub-systems such as IFB for commentary and cueing. And as you may have experienced, most facilities end up using a combination of these different options, so the total number of possible outcomes is very large.

A challenge in understanding intercom functionality and operational capability is the confusing use of identical terms that have different meaning in different contexts. To simplify this guide, we are going to suggest looking at the two most used communication protocols between individuals as either a direct path between two individuals called a **"point-to-point"** or as communications among a group of people often referred to as a **"partyline"**. In some applications and/or in specific countries a partyline is called a 'conference'.

There are several other forms of intercom communications that can be considered after a basic design including:

a) *Group Calls* (sometimes noted as 'one-to-many') includes a set of participants or interfaces such as controls, interfaces, key-panels and partylines that can be treated as a single item. A user who has a talk key programmed for a fixed group can simultaneously talk to everyone who is part of that fixed group. A fixed group differs from a partyline in that the group membership is set by the configuration of the intercom system (not changed dynamically). For example, a Group Call can be an "all" call from a director to get the attention of everyone involved in the production.

b) *ISO (isolated) Calls* – permits a user on a conference, (partyline) to be isolated from that conference for a private conversation. ISOs are typically used by a video engineer to isolate a camera from the camera Partyline

c) One-way communications called *IFB (interruptible foldback)* for cueing. These systems are used to cue talent. An IFB can be provided to people other than those designated talent and it can also feed a loudspeaker (such as in a dressing room) as well as an earpiece. A typical example is a background music system that can be interrupted with an announcement.

SECTION 1

COMMON COMMUNICATION TECHNIQUES

The most common method people communicate operating an intercom is, by far, the Partyline format.

1.1 Partyline

The term partyline can refer to any of the following:

- A wired two-wire intercom system developed by Clear-Com
- A wireless intercom system
- A conference on a two-wire system
- A conference on a matrix Intercom system

Technically speaking, a partyline intercom (also referred to as 'talkback' systems and two-wire systems, in some parts of the world), is a communications system where the path is the same for both talk and listen. An analog partyline intercom circuit transmits and receives audio on two-wires. Digital and/or network partyline intercoms perform the equivalent functions except they are Ethernet based.

This system design is 'conference by nature'. 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, despite the format, are full-duplex, (one can hear and talk at the same time) 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. For example, one person can talk, while all the others on the bus or channel can hear. Most users only converse on one or two channels, receiving cues and talking within their own group.

> Within the audio industry the name "Channel" can also be used to describe A wireless beltpack, an audio path in an audio console, a 2-way radio or a wireless microphone.

Keep all of this in mind about partylines because as a design criteria later in this book you may ask yourself, how do the people in my facility function and, are they best served to communicate as a task-specific group?

"PL" is an abbreviation for "partyline" but in the vernacular can refer to an intercom system, a user station on an intercom system, as well as any kind of communication conference (including a teleconference). With partyline systems, the talk and listen path is almost always associated with and defined by the term – **'channel'**.

For instance, there could be a front of house audio mixer, and an on-stage audio mixer, several audio technicians, and a person responsible for all the wireless microphones in a facility or show. This is a good example of a group of people who share a common workflow, i.e., audio. An appropriate intercom design would include supplying these users a partyline circuit and this partyline designated and called the audio 'channel'. (Be aware that other users can or may need access to any or all the audio personnel – as an advanced design requirement we will cover this in the design section.)

Figure 1 is an example of a classic partyline system:

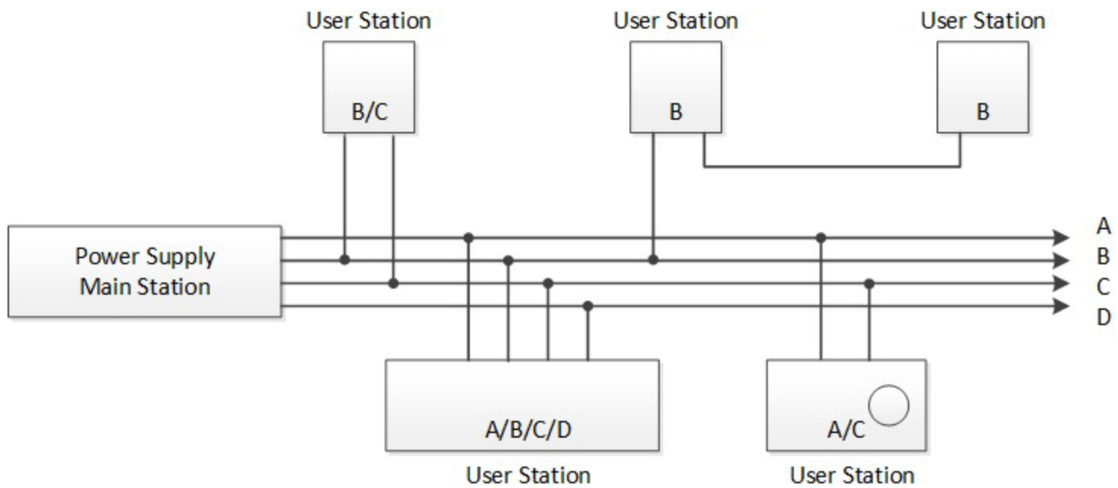


Figure 1: 4 - Channel Clear-Com Concept using Single Twisted Pair Shielded Cable

A classic 'analog' partyline flow will always include a simple power supply or powered main intercom station used to 'wet' the intercom line circuits. 'Wet' refers to the detail that there is a DC Voltage applied to the line to power the user stations. The audio i.e., the talk and listen are carried on the circuit as well. This simple system in Figure 1 includes at the main station/PSU 4x partyline 'channels' we see labeled A/B/C/D. In a real application these 'channels' might be given to a group of users and labeled as such – an example looking at Figure 1 would be to assign Channel B used by the group of Audio staff working in the facility.

Recognize that the four wet channels are distributed throughout a facility and the end user stations connect the appropriate channel circuit as required.

This would be a good point to explain that with static analog partyline systems each channel is a separate wire. With digital partyline systems or IP network partyline systems, all the channels created within the system are available on the one wire and system configurations define which channels are available at each endpoint within the system. That flexibility allows the user selectivity of which channel(s) they talk and listen on. More of this is explained below on getting down to choices between analog systems and digital systems and infrastructure decisions. Conference workflows are not limited to stand-alone analog or networked partyline systems but can also be a configured workflow in the more powerful digital matrix intercom platform and added to user stations as required.

As a reference to be discussed later, for partyline/conference operation, we need to know how many simultaneous conversations need to be supported. This will tell us the number of channels required for either analog or digital partyline options.

1.2 Point-to-Point (PtP)

A point-to-point workflow allows a one-to-one communication path. Within production intercoms, a Point-to-Point is a direct communication between two intercom stations or between stations and interfaces. It is the basis of communication in matrix intercom systems. Conversation is limited to those selected by the originator of the call thus provides complete privacy between stations. This connection normally includes a "tally" (call light) subsystem to alert the destination of the caller. Most connections are full duplex. A common example is a point-to-point configuration between a producer and director of a show. Note: With large multi-channel partyline systems it is quite common that a partyline channel can be used as Personal Party-Lines (PPLs) between two users for simulated PtP communications.

Starting with a simple design is preferred. As stated earlier, the majority of intercoms are simple wired partyline systems with the addition of wireless partyline intercom. This is based on several design criteria that the designer may consider such as the technical ability of the user to use an intercom station and/or the frequency that an intercom may be used versus the cost for wiring a facility.

1.3 Wireless Intercoms

For most applications, the wireless intercom is an extension of the wired system, used by those staff members who require mobility for safety or convenience. Wireless intercoms are typically used in a group/partyline communication workflow as detailed in the analog party-line section above. They are seldom used as point-to-point communications; however, there are wireless systems available that have these capabilities and are detailed later in this guide.

Systems are generally comprised of a "base station" and a limited number of user wireless beltpack/headset stations that can work with and are 'paired' to the base station. Wireless intercom communication is full-duplex; however, some systems offer different operational modes whereby the system can be used in a half-duplex manner, allowing more wireless users on the system.

1.4 Pre-planning Notes

A typical wired or wireless production intercom system will always consist of Users, sometimes IFB Circuits, sometimes Cameras, and most often miscellaneous ports (to include, 2-way radios, analog & digital telephones, paging, other intercom interfacing, etc.). The first step is to determine the size of the central intercom by counting everything that is attached to it.

Users

The users of the intercom are operators with keypanels, speaker stations, wired and wireless beltpacks or (most recently) software clients on PC's, mobile phones and tablet devices.

IFB Circuits

If IFB is considered necessary, we need to add the number of IFB circuits, i.e., as described above in the sidebar, the number of users who receive cueing. Sometimes cueing talent can be more general rather than individual and thus be regarded as paging. See 'paging" below. Within partyline systems the need for IFB circuits is often satisfied with a separate IFB sub system.

Cameras

Should cameras be in the facility the easiest and simple solution is to count the number of cameras as individual circuits. It is not advisable to ever daisy chain camera circuits or to bridge them without proper mixer circuitry. Camera circuits can be combined within the intercom in to a partyline channel if it is required to meet a workflow. The best way to interface a camera intercom to an intercom system is, without exception, to access the Camera CCU intercom circuit as a four-wire circuit. A four-wire connection is frequently not available on lower cost cameras. For these projects we often recommend running a microphone cable with the camera cable and plug a beltpack in at the end or consider a wireless beltpack system.

Paging

Paging is a one-way or simplex communication circuit from the source (or originator of the page) to the destination. A few examples for paging destinations may be stage or studio announce, dressing/green room pages, lobby pages and back-of-house pages.

Classic and Ageless

Should you completely replace your existing intercom system? Possibly not completely! Clear-Com has been around for a long time (over 50 years) and there are systems that have been in constant service for almost that long. The good news is the newest analog two-wire partyline products are completely compatible with the older two-wire analog partyline products. In fact, all Clear-Com products, analog or digital or IP based can be interfaced with legacy analog partyline equipment. A cable refresh and general troubleshooting review is a good place to begin if keeping equipment as part of a system expansion.

SECTION 2

CHOICE OF EQUIPMENT TYPES

2.1 Overview

Currently available analog two-wire partyline systems offer high performance, low to moderate cost and can use one channel or multiple channels. The advantages of two-wire partyline systems are simple wiring, easy expansion to additional stations, very little central equipment needed, generally a low cost per station, and simple operation for team/group type activities.

The disadvantages of two-wire systems are their reliance on hybrids (two to four wire converters) which are required for interfacing between systems and cause significant degradation of system performance, the difficulty encountered in interfacing them to other two-wire systems that have different characteristics, their lack of selective calling to multiple stations, and their limited privacy capability. Within applications where intercom allocations often change a big disadvantage of analog partyline intercoms is that changes to distribute/route channels require re-cabling, although there are some ways to reduce the cost/complexity involved – please speak to your Clear-Com point of contact for more details.

Digital network partyline systems also offer high performance, can use one channel or multiple channels, and offer fast reconfiguration changes of routing channels amongst users but come with a much higher cost. The biggest advantage of the digital partyline systems is simple wiring, i.e., compared to analog systems which are wired per circuit, with digital systems all channels are on one circuit/wire. These systems are Ethernet-based and therefore take advantage of the flexibility offered in IP networks.

Matrix intercom systems have numerous advantages over other types of intercoms.

These advantages include size, configurability, variety of communication types supported, and ancillary functions available. The most simplistic matrix (four-wire) intercom circuit transmits audio on one pair of wires and receives on a second pair. This format is point-to-point by nature and can be pictured as a star configuration – each station connects to the center through its own multi-conductor link. Today these are IP network-based links which, while adding the complexity of Ethernet networking, remains the same for feature, function and usability. Instead of subsystems to achieve different functions, the central processor and software permit the system to be dynamically configured for different forms of communication.

Since each user station has the electrical capability to be connected to any other user station (via the cross points or individual crosspoint adjustments), changing who talks to whom, rules for what happens under certain circumstances, and the assignments of keys are under software control.

The noted drawbacks to matrix systems are that they can require substantial central equipment, are expensive, generally require more wiring than partyline systems, and can be expensive to expand.

Whether you use two-wire, wireless, or matrix-based intercoms the most recent developments unite these platforms in a single system, self-contained, IP cross functional central station. Clear-Com's Arcadia Central Station is an excellent example of this kind of unified system.

A central station can bridge from analog to digital and connects both wired and wireless intercom products. This offers best of all partyline functionality along with the dynamically configurable routing a matrix might provide. Licensed ports may be allocated freely to any port hardware type supported by the unit with no adapters or special boards required. A graphical user interface can manage common operations directly on the front panel.

Choosing an intercom system for a given facility is a blend of budget, existing wiring considerations, and potential intercom size. Complexity is quite often the major negative to matrix intercom systems. Complexity brings a whole world of issues, which can be of major consequence.

For projects that are especially intercom intensive, normally either matrix or hybrid systems that includes matrix and other interfaces, we advise that you contact Clear-Com directly for assistance.

2.2 Choice of Equipment Types - Wired Products

> Complexity, in particular, is the biggest disadvantage to matrix systems making them unsuitable for many applications.

2.2.1 Analog Partyline

Clear-Com Encore®
- Main Stations
- Intercom Power Supplies
- Remote Stations
- Wired Beltpacks
- Speaker Stations

A basic analog partyline intercom system consists of a single or multi-channel Main Station (e.g.: MS-702/MS-704 & SB-704), or Power Supply (e.g.: PS-702/PS-704) connected to various multi-channel Remote Stations (e.g.: RM-702/RM-704) and/or single or multichannel beltpacks (e.g.: RS-701/RS-702/RS-703) and/or loudspeaker stations (e.g.: KB-701/KB-702), interconnecting cable, headsets, panel microphones or push-to-talk microphones.

Clear-Com stations are interconnected with two-conductor, shielded cable, like a microphone cable, (or individually shielded multi-pair cable as required). Portable stations are interconnected with two conductor shielded cables terminated with 3-pin XLR connectors.

Large systems can be built using multiple power supplies or main stations that supply power to the intercom line(s).

Key Feature Summary:

- The power supply (normally centralized) generates DC power for the entire system (exception: self-powered user stations).
- User stations connect to the power supply and intercom line(s).
- For a given channel, user stations can be and often are connected in parallel.
- The interconnecting cable for most analog partyline intercoms is standard microphone cable.
- Distributing the amplifiers allows for better performance and more features.
- A single channel analog partyline belt pack has an intercom line connector, a headset connector, volume control, and a talk or microphone on/off switch. A two-channel belt pack adds a channel selector or two talk switches and two volume controls.
- A speaker station usually can be a headset or a speaker station.
- Speaker stations add a power amplifier, speaker, and speaker on/off switching to the headset station electronics.
- Analog partyline Main Stations are multichannel and allow a director or lead person to have separate conversations with various crews in any combination. Main Stations often have many additional functions.
- Headphones range in impedance from 50 to 1000 ohms. The headphones should also have at least 20dB acoustic isolation for concerts and athletic contests.
- Since the power supply has a limited number of connectors, splitter boxes are needed to expand the number of user stations in a system.
- Some stations have loop through connectors to allow daisy chaining of stations.

2.2.2 Network Partyline

Clear-Com HelixNet® Digital Partyline
- Main Station
- HelixNet User Devices
  - Remote Stations
  - Wired Beltpacks
  - Speaker Stations

The HelixNet system transports the intercom audio and signaling as packetized Ethernet. 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.

Ethernet based devices such as beltpacks, speaker stations and remote main stations can be connected to this powered 'powerline' output using standard twisted pair shielded, 3-pin XLR, cables, (microphone cable), using the same topologies utilized in analog Partyline systems, including passive splits, daisy chains and extended cable distances. The most important attribute to the HelixNet platform is the familiarity for previous users of Clear-Com analog partyline, while still offering impressive flexibility through configuration-based architecture.

Unlike analog partyline systems where the number of channels required must be cabled to a dedicated power source, the number of partyline channels are virtual and created in a HelixNet Main Station. The system creates 12 partyline channels with option to increase this to 24 partyline channels thru the purchase of an upgrade license and all channels share one cable.

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" supplied within a main station and comparable to analog PL and as well 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.

- For a given channel, user stations can be and often are connected to a PoE port of a network switch.
- The interconnecting cable for most digital network partyline intercoms is category cable.
- A two-channel belt pack has an RJ45 network intercom line connector, a headset connector, channel selector or two talk switches and two volume controls.

- A speaker station usually can be a headset or a speaker station.
- Speaker stations add a power amplifier, speaker, and speaker on/off switching to the headset station electronics
- Main Stations are multichannel and allow a director or lead person to have separate conversations with various crews in any combination. Main Stations often have many additional functions

Clear-Com HelixNet Digital Partyline systems include an internal web browser configuration tool called the Core Configuration Manager (CCM™). The CCM facilitates a quick and simple means of configuring any devices in a LinkGroup, including role-based configuration of endpoints, account management, save-and-restore, and live monitoring of all system components.

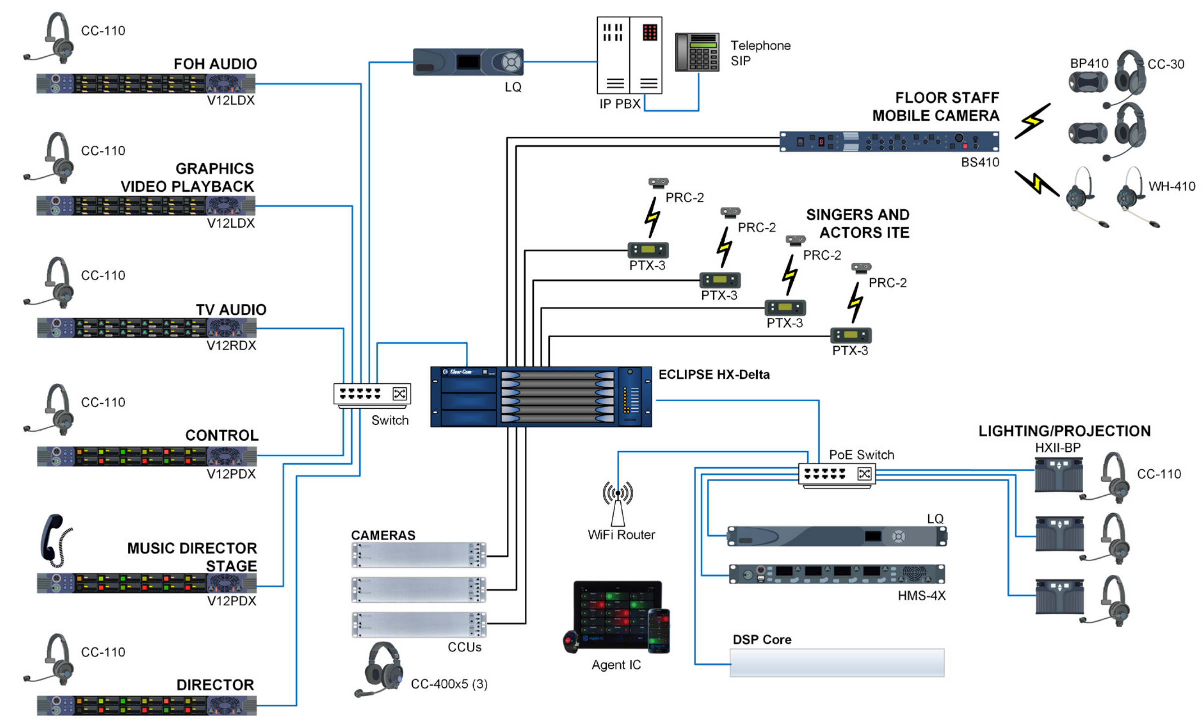
2.2.3 Matrix

Clear-Com Eclipse® HX
System Frames
Interface Cards
Interface Modules
User Keypanel Stations

The best method to understanding any matrix production intercom is from an audio perspective. These platforms are referred to as full summing, non-blocking audio matrices where a central piece of equipment ('the matrix') performs this functional task. This refers to the capabilities that any input or number of inputs could be routed to any output or any number of outputs with full cross point level control/adjustment. The term cross point refers to a one-way audio path from one ports' input to another ports' output. Cross points exist between every pair of ports in the system and are connected and disconnected as needed to provide communication paths between system ports on the TDM backplane. The outstanding feature here is the cross-point level control/adjustment, which is not typical of traditional audio routers. Clearly, when you have many people talking, the ability to adjust listening levels (cross points) is critical to production cueing and directions. Configuration software is used to create and manage communication pathways between devices including matrices, interface cards, interface modules, and user subscriber keypanel stations.

The basic hardware elements of Eclipse HX Matrix Intercom systems are:

- The Matrix Card Frame – see definition above. Characteristically this frame retains the interface modules and circuit cards although ins some cases these may be integral as opposed to hot-swappable.
  » The heart of any Matrix intercom must have a Central Processing Unit (CPU). A CPU stores complete system configurations in its memory. Each matrix system contains at least one central processing unit. Most if not all matrices are supplied with two CPU cards (in a primary and secondary relationship), ensuring fail-safe operation.
- Various circuit cards, sometimes called client cards, used for I/O
  » There are a wide variety of different client cards available for matrix systems. These client cards are the I/O and interfacing connections to various user intercom stations and third-party sources and destinations. These client cards offer various audio format conversions as well. Client cards are described in terms of "ports." Ports refer to the number of connections available to external devices to and from the matrix although with IP based cards these ports are virtual. Typically, a port is assigned a function within the configuration software. This function specifies what type of device is connected to the port such as a user key panel, an interface, or a four-wire. Software is used to set the parameters of the function for that port.
  – Rear-panel Connectors. A matrix connects to remote devices such as user key panels, interfaces, general purpose inputs and outputs, local area networks (LANs) and other matrices through its rear-panel connectors. RJ45 connectors are found on analog four-wire, IP circuit cards, and Ethernet for PC control, while digital connections may include BNC or fiber or both for MADI I/O. Typically D-sub connectors are used for externals functions like GPIOs.
- Interface Modules
  » Because the intercom is considered 'critical communications' each matrix frame typically operates with two independent power supplies. These must be connected to a main and backup power source for redundancy. If one of these power supplies fails, the second supply automatically takes over.
- User Control Panels
- An external computer (PC), which hosts the configuration software



This drawing represents the variety of connectivity options available through the Eclipse matrix system.

Matrix Trunk Systems

There inevitably comes the need for users in one matrix intercom system to communicate with users in another. When the facility has grown and thus the need to connect multiple matrix intercoms that serve different areas within a given facility or separated by countries, we employ trunking methodology. Clear-Com trunking schemes are referred to as "intelligent," meaning it functions automatically, establishing the trunk, continuously monitors and reports on status of trunk utilization, and releasing the trunk when the conversation is over.

There is a number of methods for interconnecting multiple intercoms using digital circuit cards, E1/T1 with E and M signaling, IP circuit cards, and optical fiber circuit cards in order to provide varying degrees of fault tolerance and redundancy. Some of the high-density circuit cards like MADI and IP may be partitioned to allow multiuse, such as X number of ports for user stations and X number of ports for trunk circuits, allowing flex-ibility in design and affordability.

In general, as one plans for expansion, you need, as a maximum, as many trunk lines as there will be different conversations.

EXPERT OPINION

Clear-Com Digital Matrix intercoms are highly configurable and customizable solutions with many complex options. We encourage you to contact the Clear-Com support channels mentioned at the end of this book for design assistance.

2.2.4 Central Station

Clear-Com Arcadia

A Central Station is the newest generation scalable IP platform that integrates wired and wireless partyline systems. With a mixture of 2-wire and 4-wire audio ports, the 1RU device serves as the core of an intercom system that includes Clear-Com endpoints along with third-party Dante and AES67 AoIP devices up to 96 licensed ports, A Central Station supports the entire range of FreeSpeak wireless solutions in 1.9/2.4 & 5GHz.

Users increase the number of total ports from 32 to 96 – with port licenses available in increments of 16. Ports are considered as: A wireless beltpack occupies one port. A Dante stream channel occupies one port. A HelixNet PL channel occupies a port (up to 24 PL channels total).

A Central Stations includes a CCM easy-to-use browser-based software tool just like the browser in the digital networked partyline, LQ and standalone FreeSpeak wireless base stations. Configuration menus enable quick set up, configure and edits to your system. It provides an overview of all the system components and interfaces and status monitoring of all the resources that are connected to the Central Station. These include wireless transceivers, beltpacks and interface ports.

2.3 Choice of Equipment Types - Wireless Products

2.3.1 Wireless Intercoms

FreeSpeak™ Digital Wireless

- FreeSpeak II® Base Station
- FreeSpeak Edge® Base Station
- FreeSpeak™ Wireless Beltpacks
- FreeSpeak Transceivers

The wireless intercom is an extension of the wired intercom system, used by those staff members who require mobility for safety or convenience.

> **Policies:** The use of radio equipment is subject to regulations in each country. A given device may not be allowed to cause harmful interference to other authorized users. The device must accept any interference caused by other users. To comply with FCC part 15 rules in the United States, radio equipment must only be used in systems that have been FCC certified.
>
> RF equipment must be installed by qualified professional personnel. It is the responsibility of the installer to ensure that only approved equipment/systems are deployed and that effective radiated power does not exceed permissible limits established by the country's regulatory agency in which it is used.

Clear-Com wireless intercom systems are generally comprised of a rack-mount Base Station and a limited number of user wireless beltpack/headset stations that can work with and are paired to the base station. The IP-enabled 1.9 GHz and 5 GHz transceivers that are functional with the FreeSpeak Edge Base Station can be used as part of Clear-Com Eclipse HX matrix systems when the matrix includes an IP interface card, the E-IPA card.

Modern FreeSpeak digital wireless systems include wired intercom features such as call signals and remote microphone kill. These modern digital systems also offer multichannel wireless user belt packs with two, four, and up to eight channels of party-line and/or point-to-point communications. Wireless intercom communication is full-duplex.

Advantages of wireless intercom systems include:

- Greater freedom of movement for the user
- Simple installation; avoidance of cabling problems common with a wired intercom
- Reduction of cable "trip hazards" in the performance/work space

Wireless Technology

Clear-Com offers nine (9) distinct wireless intercom solutions starting with a basic 2.4 GHz-based radio system that does not interface to any other external intercoms or sources, going all the way up to extensive FreeSpeak systems that can be customized per application. In between are various options that offer different feature sets and price points.

The frequency spectrums used in Clear-Com wireless solutions include 1.9 GHz DECT, 2.4 GHz and 5 GHz radio technologies.

The most popular solution is the 1.9 GHz FreeSpeak II system which is based on dedicated distributed wired transceivers. From an RF installation point of view this FreeSpeak system can be considered as a plug-and-play system. There is no frequency coordination and the installer simply needs to install the transceivers in pre-determined coverage zones and with proper install methods as explained in the manual. For systems that use individual transceivers per base station, the transceivers of each individual system should be separated from every other system's transceivers for best co-located performance. These transceiver distribution systems carry FCC licensing.

Clear-Com FreeSpeak Base Station systems include a browser-based configuration tool called the Core Configuration Manager (CCM). This CCM facilitates a quick and simple means of configuring wireless beltpack devices including role-based configuration of endpoints, account management, save and restore, and live monitoring of all system components.

Transceiver Placement

One of the most important factors in a successful RF system is the placement of the transceiver relative to the desired coverage area. Digital wireless intercoms (2.4 GHz, 1.9 GHz DECT systems and 5 GHz) use a dual transceiver diversity system. It is critically important that transceivers be connected and properly placed at all times for best RF performance. Unlike older analog VHF and UHF wireless intercom systems that had dedicated transmit and receive antennas, each digital transceiver is both a transmit and a receive transceiver.

Some important points to remember when placing DECT transceivers include the following:

- Every transceiver has a certain pattern of coverage for which it is useful. The patterns of 1.9 GHz DECT system transceivers need to overlap in the desired coverage area to ensure best RF results.
- Higher is almost always better when placing antennas. In many cases, the best placement is to get the base station transceivers well above the desired coverage area (at least above head level) and point transceivers directly down at the coverage area.
- Maintaining a direct line of sight from the base station transceivers to the belt pack is the best possible scenario for signal strength.
- Always keep transceivers away from:
  - » Large metal objects; stay at least two feet away.
  - » Large containers of liquid. Most liquids absorb RF.
  - » Confined spaces. Do not set up transceivers in rooms or areas that are closed in with very few RF enter/exit points.

Interfacing to Wireless Partyline Systems

2W/4W Interfacing. Wireless stand-alone intercom systems (i.e. not within a matrix frame) are commonly equipped on a per channel basis with a two-wire loop-thru (three-pin XLRM&F) interface to analog wired party-line systems and analog line level four-wire I/O (typically an RJ45) for connection to digital wired party-line systems, matrix intercoms and other third-party audio devices. The two-wire interfaces generally have options for interfacing to both RTS TW and Clear-Com party-line systems. Send and receive audio trims are incorporated for both 2W and 4W I/O. All base stations include a nulling feature when connecting the analog wired two-wire circuit.

2.4 Choice of Equipment Types - Intercom and IP Networks

Communicate Over IP

The days when real-time audio and IT networks made poor bed fellows have surely come to an end. The attraction of IT networks as an audio carrier has never been clearer: low cost per connection, very high capacity, flexibility, and an established global infrastructure.

Transmitting quality intercom signals over networks is not without its challenges. IT technology has not traditionally supported synchronous real-time traffic, and typically uses retransmission to compensate for underlying unreliability—a technique wholly unsuited to voice communications. However, higher bandwidth connections, advanced clocking protocols with sophisticated VPN management tools allow standard networks to rise to the challenge of routing intercom on a local and global basis.

For IP based digital audio intercom systems, there are typically four scenarios that benefit from IP-based connectivity:

- Operator key-panel-to-host matrix/main station (including soft clients)
- Matrix-to-matrix or Central Station intelligent linking
- “Glue” systems that interconnect traditional systems over IP infrastructure (LQ)
- Matrix/Glue Systems-to-digital telephony gateways

Central to the use of off-the-shelf IT components is conformance to a set of standards which, together, define IP networking. This includes protocols such as RTP, IGMP, QoS and PTP, all of which are used in audio and video over IP streaming but would be familiar to IT specialists outside the broadcast industry. (See IP Glossary)

In addition, take notice that the production intercom, particularly in the Broadcast and AV media market segment is moving into a new way of working, away from dedicated 1:1 cable connections between equipment, to using Audio over IP (AoIP) networks using the AES67 digital audio or SMPTE ST2110-30 standard which includes AES67 digital audio.

Consequently, **the success of the installation and project relies on a correctly configured IP network infrastructure**. There are some practical steps you can take in pre-sales and installation phases to make the install go smoothly.

RECOMMENDED

We recommend that for any installation you have a network engineer available onsite from either your company or the ethernet switch supplier who can setup the equipment correctly.

IP AES67 AoIP

What Is AES67?

- It is an High-performance Streaming Audio-over-IP interoperability standard
- It is for audio transport only
- It is not a complete system. AES67 is a feature or option in a wider audio system which can fulfill other tasks such as routing, monitoring, discovery, or system control.

AES67 uses the IEEE 1588 Precision Time Protocol standard for synchronization.

AES67 data packets are IP packets formatted according to the Real-time Transport Protocol (RTP). The RTP standards define packet formats for numerous types of audio and video.

AES67 operates over standard layer 3 Ethernet networks and, as such, is routable and fully scalable.

AES67 does not include specific requirements for discovery and control functions. Products can use solutions of their choosing for discovery and control.

Interoperability without discovery or control is ensured by mandating use of SIP for connection management and designating the SIP URI or SDP description as the information that needs to be distributed through a discovery system. Standard use of a few discovery systems including Bonjour and SAP is discussed informatively in the standard.

Clear-Com AES67 enabled devices use a proprietary common discovery method.

Which Clear-Com devices broadcast AoIP AES67?

- The Eclipse HX Matrix E-IPA-HX interface card
- The Eclipse HX Matrix V-Series Iris Panel
- Clear-Com IP wireless transceivers (IPTs):
  - » FreeSpeak II: FSII-TCVR-IP-19 (1.9GHz)
  - » FreeSpeak Edge: FSE-TCVR-5-IP (5GHz)

IP SMPTE ST2110-30

What Is SMPTE ST2110-30? This suite of standards (ST-2110) specifies the Real-time Transport Protocol, (RTP-based) transport, synchronization and contribution of separate video, audio, data streams over IP networks. The ST2110-30 describes RTP based PCM digital audio only + SDP metadata (RFC4566) for stream reception and interpretation over IP networks by reference to AES67. An SDP-based (software-defined perimeter) signaling method is defined for metadata necessary to receive and interpret the stream. Be aware of differences within 2110-30 and pure AES67 domain.

RECOMMENDED

For a successful ST2110 installation with your Clear-Com system, you should have a technical and competent network engineer available onsite who knows how to configure the network switches and understands the requirements of AES67/ST2110 systems. Note: An AES67/ST2110 network is not the same as a standard computer IT network. An incorrect setting on a switch can stop data or flood the network with data.

If your project is determined to follow SMPTE ST2110 standards we recommend you call Clear-Com directly for advice. This booklet does not detail design for intercom within SMPTE ST2110 facilities.

We do offer several white papers towards Network Guidance:

AoIP Networking Guide

Networking Guide for FreeSpeak Edge Base Station and Arcadia

HelixNet Networking Guide

2.4.1 Virtual Intercom Clients

Agent-IC® Mobile App

Station-IC™ Virtual Desktop Client

The concept of virtual intercom was inevitable with the proliferation of the use of IT networks, and necessary to support this challenging paradigm shift to ensure a smooth transition into the new world of remote live event production and broadcast-IT. Selected other market segments are finding this decentralized workflow more and more applicable to their needs as many job functions are being moved to remote or remote-optional.

User PCs/tablets/smartphones as we call, “soft clients” can be easily located anywhere in the world as part of a network infrastructure where there are multiple facilities, control rooms, studios, edit bays, audio sweetening, multiple wired or wireless IFB audio channels, ENG, SNG, and full production trucks. Features of virtual intercoms include:

Multiple conference modes, including talk only, talk and listen, or listen-only mode capabilities. One-to-one, one-to-many, partyline, ISO and IFB communication with individual listening levels in addition to call signals and logic controls are supported.

Clear-Com implements a fully-featured mobile intercom client from the respective IP circuit card in the Eclipse HX hardware matrix platform. Through the use of various LQ gateway interfaces, soft clients can communicate with hardware matrix key panels and once introduced to the matrix, can be programmed to any and all elements within the matrix environment.

Our software app, Agent-IC, utilizing hardware-based system resources, is available for mobile phone and tablet devices thru the phone app store. Station-IC is a virtual desktop client facilitating a scalable intercom user station and can be freely download from the Clear-Com website.

2.4.2 LQ Series of IP Interfaces

LQ Series of IP Interfaces

LQ interface products can be considered as gateway IP enabled bridging devices meant to transport, distribute, and merge audio and communications signals such as analog 2-wire partyline, analog 4-wire audio or analog 4-wire audio with GPIO over LAN, WAN, or Internet IP infrastructures. You can network geographically-dispersed audio and communications devices of any brand or technology type with an LQ device.

LQ is administered using an integrated, browser-based software utility called the Core Configuration Manager (CCM). It facilitates quick and simple configuring any device in a Link-Group, including role-based configuration of endpoints, account management, save and restore, and live monitoring of all system components.

LQ Series 4.0 and above offers a mix of physical and 'virtual' ports. All LQ devices include SIP connectivity to external telephone lines into your LQ or LQ link group intercom system. All LQ devices also include access to license enabled Clear-Com mobile client, Agent-IC, and Station-IC for PC.

LQ also offers network system expansion connectivity and the ability to configure a Link-Group to HelixNet network partyline systems. An LQ/HelixNet link group provides HelixNet to IVC connectivity to connect to a Clear-Com Eclipse matrix system. An LQ device and HelixNet product families share SIP/VoIP telephony lines and the Clear-Com mobile client, Agent-IC, and Station-IC for PC licenses within each LQ.

Meshed Wi-Fi networks are typically used for local communications with 3G, 4G, LTE networks used for remote clients.

It's said that an experienced group of people can usually communicate pretty well without having to say a word. If you have completely read all of the previous material congratulations! But you might be slightly overwhelmed because if you are not designing intercoms often then this info might feel like taking a drink from a fire hydrant. Very hard to get it all in one reading with so much information. Especially if intercoms are new to you.

Summary

Intercom systems are often misunderstood and considered an oddity. Partyline intercom systems are the simplest form of this important solution for live event or simple television production communications. Digital intercom systems have become increasingly powerful, flexible, and complex. This is particularly true for matrix and wireless intercoms. The size and features of modern intercom systems allow more users on a communications system and high-quality communication between those users can take place from disparate and far away locations. The end result is worth the added effort as these communication systems have helped make increasingly complex broadcast productions and virtually produced theatrical productions and corporate events possible. The melding of IP technologies with production intercom has yielded a variety of solutions that can create a local or global intercom network.

It is time to move forward to the design process starting with understanding the user needs.

SECTION 3

DEFINING YOURE NEEDS

3.1 User Details

Questions to ask:

- What is the application or venue? (Live event, broadcast, sports production, theater)
- Who are the operators and who talks to whom? Are there any specific restrictions?
- For conference operation, you need to know how many simultaneous private conversations need to be supported. This will tell you the number of channels required.
- How many fixed and/or seated positions? (Producer, director, stage manager, graphics, audio mixer, lighting board operator, engineering, green room)
- Who needs headsets; who needs speakers and microphones?
- Who might require a wireless user station beltpack? (See wireless section)
- What is the required station form factor for each operator?
- Are any stations at a remote location? If so, at what kind of distance – across campus or across the world?
- Does the system require interfacing with existing equipment such as an older intercom or telephone lines, radio systems, or any other outside systems? And will those be permanent integrations or temporary/event based? (See interfacing section)
- What types of headsets are required? Single muff/dual muff?
- Does future expansion need to be considered?

Whenever designing something new it is always best to follow a plan whether your needs include buying, renting, or expanding a system. So, a good question is - how many individuals and/or persons need to communicate with one another?

It may be useful to list the disciplines or departments within the facility or show such as:

- audio
- lighting
- video
- graphics
- transmission
- stage management
- production
- stagehands
- electricians
- carpenters
- cameras
- talent/announcers

If the facility incorporates multiple spaces/venues, list the departments per space individually and address any intercommunication between the different spaces later. Within these disciplines or departments list the members or people who may work here. Define crew/team needs and assignments/positions.

Create spreadsheet/example of who talks to who

A spreadsheet is a good place to start to layout who talks to who and you will quickly start to see the overall intercom layout come together.

1. Let's start with a TV/video production example.

Department/ Position	Who they talk or listen to
Example Discipline: Video	
Vision Mixer/Technical Director	Has to talk to any/everyone. During show operations, he mostly talks to: Director/A-1/tape/ EIC/ GFX/Prompter/Stg. Mgr.
Graphix	Statistician/Producer/Director/TD
Graphics coordinator	Graphix operator/Booth Stats/Official Stats/Alias (via telephone, pregame)
Shader	TD/ Director/Cameras (ISOs)
Cameras	TD/ Director/Shader
Replay	TD/ Director/Producer/A-1/EIC

Example Discipline: Production	
Producer	Talent/ Director/A-1/Tape/Floor-Stage MGR
Director	TD/ Cams/A-1/ Tape/GFX
Floor Manager/Stage Manager	Producer/A-1
Writer	Psychoanalyst
Talent	Producer/A-1
Teleprompter	Producer/A-1/Talent
Lighting Designer	TD/ Shader/Director/Producer
A-1	Everyone
A-2	A-1/Stg. Mgr./Caterer

2. Here is a theater example. PL's = "Channels" (PL's is short for partylines and a PL is often called a 'Channel'. Channels are explained in Section 1) (Pvt. = is a "Private" channel between these users)

User	Intercom Station	Ch. A	Ch. B	Ch. C	Ch.D
Stage Manager	RM-704	Deck	Lights	Spots	Sound
Automation	RM-702	Deck	Deck Pvt.		
Lighting Design	RM-704	Lights	Lights Pvt.	Spots	Spots Pvt.
Assoc. LD	RM-704	Lights	Lights Pvt.	Spots	Spots Pvt.
Asst. LD	RM-704	Lights	Lights Pvt.	Spots	Spots Pvt.
Sound Design	RM-704	Sound	Sound Pvt.	Lights Pvt.	
Assoc. SD	RM-704	Sound	Sound Pvt.	Lights Pvt.	
Spot 1	RM-702	Spots	Spots Pvt.		
Spot 2	RM-702	Spots	Spots Pvt.		
Spot 3	RM-702	Spots	Spots Pvt.		
Light Op	RM-704	Lights	Lights Pvt.	Spots	Spots Pvt.

This spreadsheet illustrates that there are 8 PL channels needed.

1. Deck
2. Deck Pvt.
3. Lights
4. Lights Pvt.
5. Spots
6. Spots Pvt.
7. Sound
8. Sound Pvt.

The **Stage Manager** justifies a requirement for a 4-channel user station because we have considered the 'workflow' includes communication with the people on deck, lighting people, the spot operators, and the audio crew. You may want to re-read the reminder about how workflows helps understand the intercom needs.

User	Intercom Station	Ch. A	Ch. B	Ch. C	Ch.D
Stage Manager	RM-704	Deck	Lights	Spots	Sound

QUICK TIP

An often-useful reminder is to consider if a person is in a fixed position, he/she should be using a wired intercom product, mobile users, such as a stage manager or set changer may need wireless, so most system designs are a hybrid of wired and wireless. In chapter 2 we discussed equipment types.

3.2 Facility Details

A somewhat complimentary method to determine intercom needs is to look at the facility itself for building-wide communication needs. This is sometimes a useful approach to spread the design needs by how many individuals and/or persons, (users) need to communicate with one another within a facility space/venue.

When you outline the facility sections you will have a list of users and then you can apply the **User Details** as described in the above section 3.1.

Organize them by logical grouping or location: (TV News)

Studio A Floor

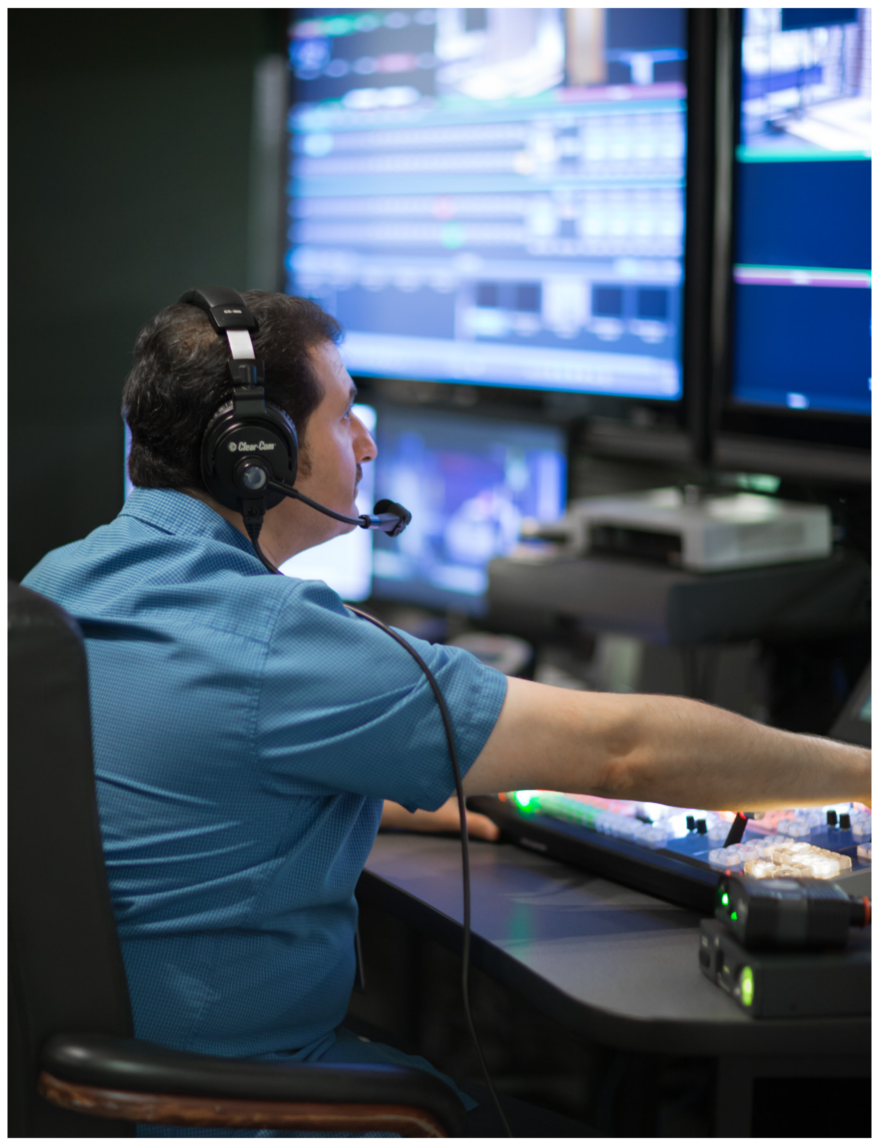
Floor Director
Lighting Director
Camera 1
Camera 2
Camera 3
Tele-Prompter
Anchor A
Anchor B
Anchor C
Weather

Control Room A

Director
Producer
TD
Segment Producer
Audio Operator
News Computer Operator
VTR

Other

Green Room
Makeup
Engineering



3.3 Wireless Intercom Details

Configuring a wireless communication system to meet the needs of a specific production or facility takes careful planning regarding features and functionality. Depending on the project requirements comprehending digital technology for the intercom and RF distribution of transceiver and antenna placements and quantity is essential.

Here is advice in the form of a few questions when gathering design information. When trying to decide what wireless intercom is right, knowing how to fit the features and functionality to the workflow is key.

Questions to ask for wireless

- Will the wireless intercom be a stand-alone system or connected to a wired intercom system? If connecting to a wired intercom system, what type?
  - » When interfacing a wireless intercom to a wired intercom consider how many common channels between the wired and wireless will be required. A simple example might be wireless users who must be in communication to the audio group who are on fixed wired intercom stations. This would count as one common channel.
  - » Wireless intercom base stations most often offer a number of methods for interfacing to other audio and intercom devices however the number of interface connections may be limited. The number of these connections are a consideration.
- Wireless user beltpacks stations offer 1, 2, 4 or 8 keysets which can be populated with partyline channels and/or can be programmed so that a key can be a direct route such as a point-to-point. How many different channels might be needed in a user's wireless beltpack?
- How many wireless beltpacks user will be supported, will the system need to expand in the future?
- Of the total number of wireless users, how many need the ability to talk at the exact same time? All can have the ability to talk, but specifically we need to know how the maximum number of people talking at the same moment (across all of the audio channels).
- Where will the wireless intercom system be used? Include which cities and describe the physical facility (or facilities) and primary/secondary coverage objectives in a typical space. These details are important for selecting appropriate system operational frequency.
- List the physical area that needs wireless intercom coverage. E.g.: multiple broadcast studios, church sanctuary, school auditorium, back-of-house
- What other wireless devices are being used in the same area and at what frequencies do they operate?
- What type of headsets are needed with the system? (Light-weight, medium, single-muff, dual-muff, low profile, etc.)

3.4 IFB Details

IFB stands for 'Interruptible Foldback'. You see it in use on TV all the time although you may not notice it. It is one of the most important functions within a broadcast or town hall and some live events. IFB is a type of simplex intercom for sending a program feed and interrupt (cue) audio on "IFB" lines for a talent to monitor. The IFB line is comprised of three elements: **Program Audio, Interrupt (Cue) Audio** and a **Dip** or **Mute** control.

IFB is a method for a producer or director to speak to the talent or a moderator with instructions, show calls, timing, or other things they need to know to stay within a time sequence. Although IFB is most often used in broadcast, it can be used in interview rooms, for law enforcement for interrogations, in houses of worship for choir and speaker management and coordination.

Those in control positions (the director, producer, or assistant director for example) control the interrupt and or announce functions via control stations. Those in receive positions (on-air talent, floor managers, studio or field crew, audience, talent, and crew in remote locations) are on the receiving end of the user station feed or on the actual user stations (talent electronics or talent station) via headphones, headsets, earphones, and / or loudspeakers. Now-a-days these can be wireless dedicated receivers or smart phones and tablets. In the middle, the central electronics unit provides all the necessary inputs and outputs, processing, switching, and power distribution.

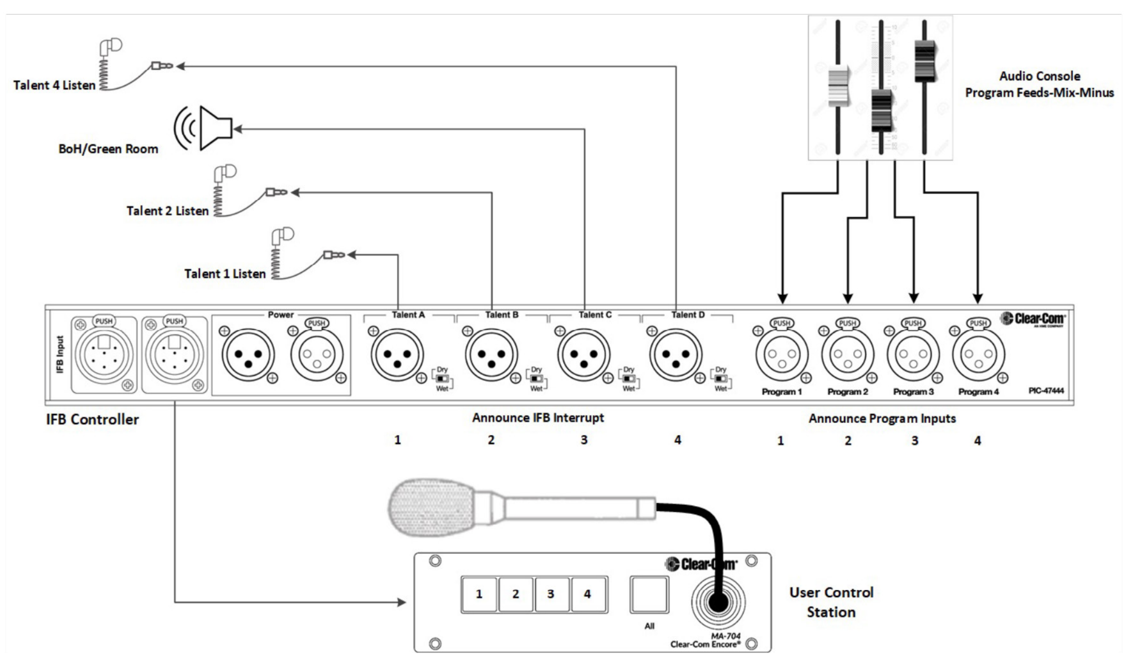


Figure 2: Illustration of a Control Position for IFB

These particular circuits are sometimes neglected or left out of the design for various reasons. Including all IFB circuits not only ensures correct intercom size, but also helps understand infrastructure topology, ancillary equipment such as any 3rd party interfacing to remote sites and the program assignment panels as described above.

Questions for IFB:

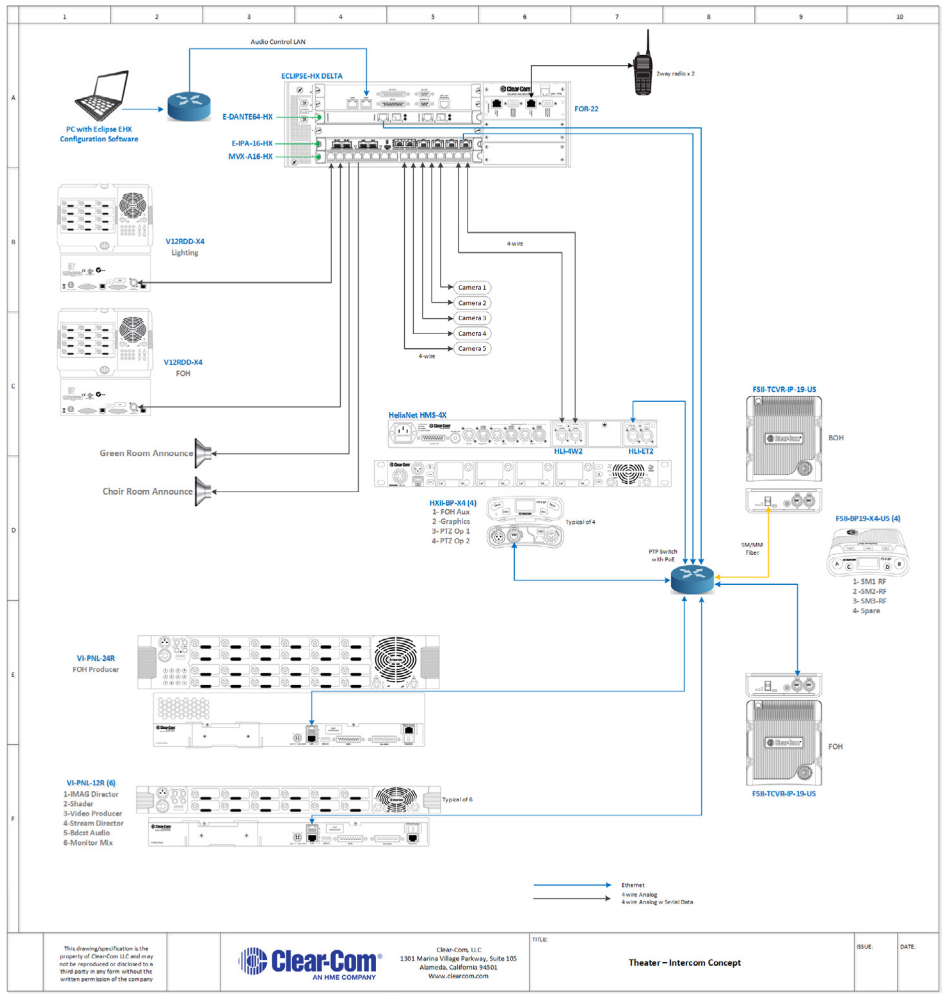
- How many talent stations must we provide for?
- How many operators need to access the IFB system?
- How many program inputs are required? What format audio might this be?
- Audio control – dip or mute – will different users require different controls and will these be changed frequently or not?
- Do any of the outputs drive radio or satellite links?
- Do any of the outputs drive telephone or other types of IP outgoing lines?
- If telephone lines and IP circuits are to be connected appropriate interfaces will be required.
- How does the talent talk back to the IFB control operator? This is often accomplished through the talents on air microphone and a pre-fader monitor tap that feeds the IFB system
- Will there be future expansion in the number of talent stations or control operators?
- What kinds of earpieces or headsets are required for the talent?
- In-the-ear earpieces are the most commonly used but headsets with boom microphones are often used for sportscaster applications.

Once you have the workflow, selecting the equipment is much easier.

Read the Blog Post "Interruptible Fold Back, AKA IFB"

SECTION 4

DESIGN EXAMPLES



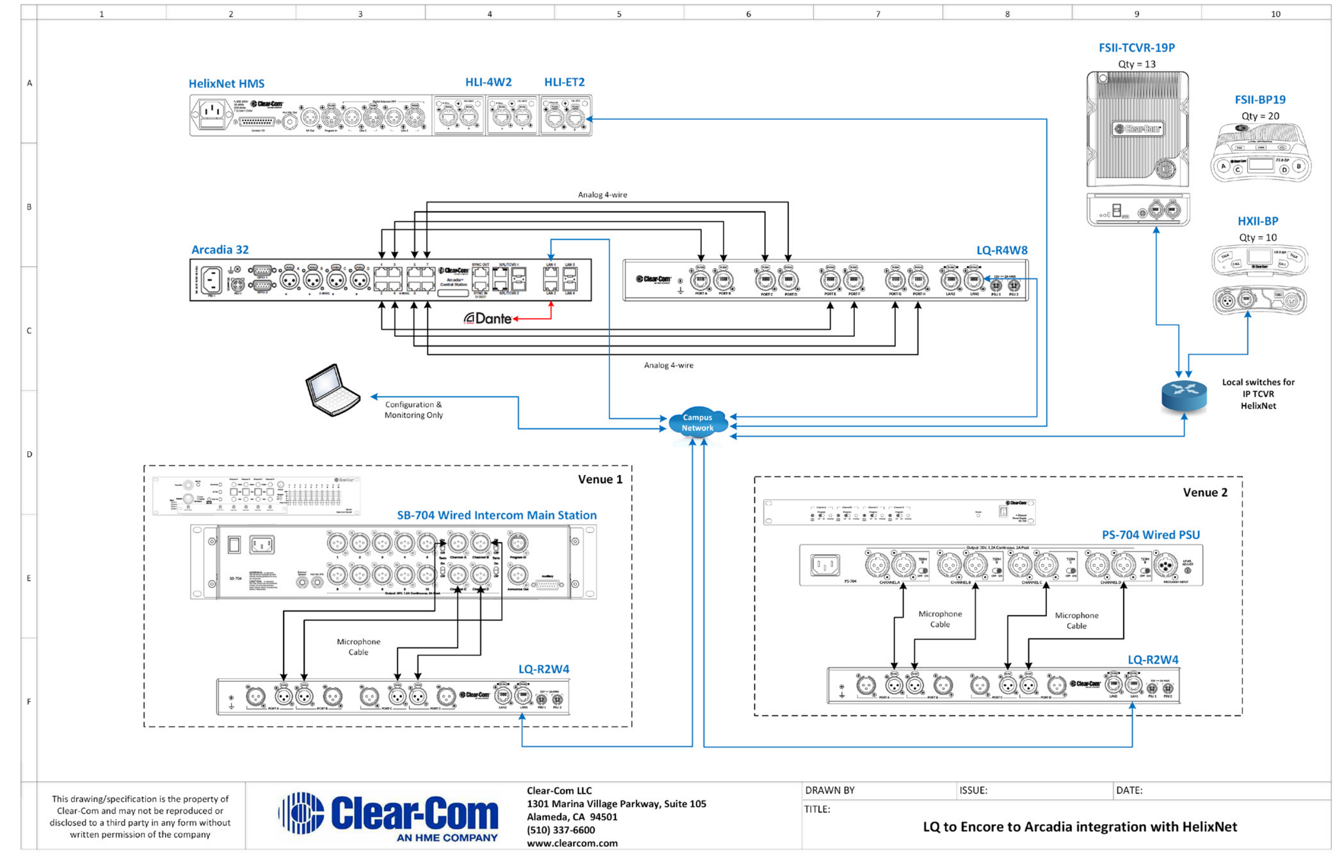
4.1 Church/Theater Using Digital Partyline and Wireless with LAN Connections

Live productions in HoW and Theaters involving elements of broadcast procedures need well-timed communication capabilities – specifically with moving talent and multiple mobile cameras. A simple way to keep the production staff in communication would be using an existing IP network infrastructure system within the facility.

How it Works

A digital matrix system hosts all connections to the wireless FreeSpeak II system via IP over the existing LAN system. All production user stations connect to the digital matrix system via the LAN network. Camera CCUs and the wired HelixNet partyline connect 4wire into the matrix system. Wired HelixNet PL endpoints are distributed over the network.

Within a configuration file channels can be created for the needed disciplines including typical, audio/video/camera/lighting/stage management workflows.



4.2 Extending Analog Partyline to Digital Partyline with LQ Series

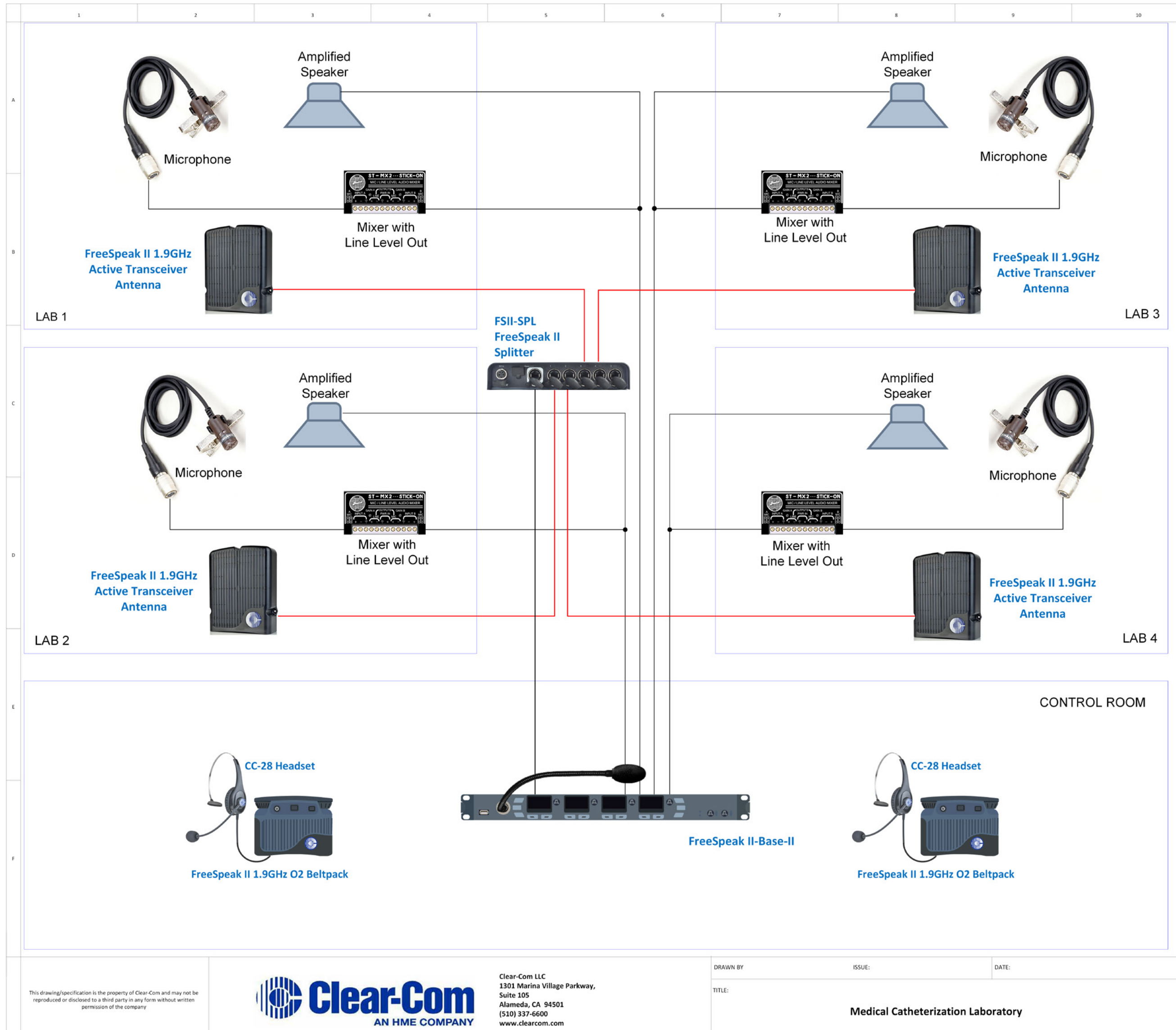
In many facilities existing analog Partyline systems, which still work well and are being used, may be repurposed when expanding adding new wireless and networked partyline. The Version 4 release for LQ Series can breathe new life into existing analog systems.

How It Works

Analog partyline systems, like Encore, can connect directly to the LQ or LQ-R device, then through IP connect to HelixNet or FreeSpeak II. In this example an Arcadia Central unit hosts the wireless and is interfaced to the wired digital partyline HelixNet with the use of the LQ. The HelixNet Main station, the Arcadia Central station and the LQ's all sit on a comms LAN network sharing their physical I/O

Application Notes

The LQ units (3 shown) connect as 'link members' to the HelixNet acting as the host. LQ interfaces connecting analog partyline system 2-wire existing systems share their resources among all LQ units on the network including the LQ at the Arcadia interfaced with four-wires.



4.3 Catheterization Lab with Wireless Intercom

A team of doctors and nurses work in the Cath Lab and an external control room. Vitals and data can be privately communicated between staff via the wireless intercom system without disturbing the patient. At any time, staff can communicate to any connected endpoint within the medical office.

How it Works

The FreeSpeak II system allows real-time wireless communication in a medical setting. From control room to Cath Lab, or any other station, staff can quickly communicate events or needs.

Application Notes

The FreeSpeak II Base II system provides four channels of communication. Through the single base station, medical staff can communicate to 25 full-duplex beltpacks or 2-wire speaker stations.

CONCLUSION

Outreach Support

We are proud to offer an Outreach Consultancy Services Program within Clear-Com. We appreciate the opportunity to work alongside you as an advisor for design on your projects that include communications. Our expertise in venue comms is extensive and we look forward to a creative and successful future co-operation.

Your Outreach contact Vinnie Macri, based in the USA, can be reached at vinnie.macri@clearcom.com. Outreach team member John Sparrow, the Head of Consultancy services in Europe can be reached at john.sparrow@clearcom.com and additional team member Peter Fong in the Asia Pacific region is can be contacted at peter.fong@clearcom.com.

As a team, we will:

- Work with you in an advisory capacity.
- Work under confidentiality and do not share your projects.
- Sit at your side to understand your project goals.
- Review your system drawings to make recommendations.
- System Concepts is always what we give, utilizing industry leading technology where relevant.
- We offer you detailed drawings of our proposed solution and a Bill of Materials.

Clear-Com solutions are available through a global network of Sales and Rental Partners, supported by Clear-Com Regional Sales Managers. Contact information for all Partners and Regional Sales Managers is available on the Clear-Com website.

GLOSSARY AUDIO/COMMS TERMS

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 truck.

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

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: is 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 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, typically may be two different signals fed to two ears independently. 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 "channel" is one individual circuit of communication used within a partyline - it is typically a two-way talk/listen path. 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 through" 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 through 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.

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: An abbreviation for Interrupted Fold-Back. It is 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.

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 path.

Linking: Linking ties separate channels into one single party line.

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".

Main Station/ Master Station: A 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 multi 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.

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. This 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.

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

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

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 Main 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.

GLOSSARY IP TERMS

Audio Format	Payload format of audio data - also known as 'encoding'.
Bandwidth	Bandwidth is also defined as the amount of data that can be transmitted in a fixed amount of time. For digital devices, the bandwidth is usually expressed in bits per second (bps) or bytes per second and expresses the amount of data required. For analog devices, the indicated bandwidth of the audio is expressed in cycles per second, or Hertz (Hz).
Bonjour	Apple's implementation of zero conf.
DiffServ	Differentiated Services- mechanism for classifying and managing network traffic, prioritization of services (e.g., low latency traffic).
DHCP	DHCP is controlled by a DHCP server that dynamically distributes network configuration parameters, such as IP addresses, for interfaces and services
DSCP	The differentiated services code point (DSCP) is a 6-bit field in the IP packet header that is used for classification purposes. DSCP is part of the differentiated services architecture.
Endpoint	A point in the system where audio data can be delivered and consumed, versus a "device" which has the capability to route audio to an endpoint.
Gateway	A default gateway in computer networking is the node that is assumed to know how to forward packets on to other networks.
Hub	Hubs are commonly used to connect segments of a LAN. A hub contains multiple ports. When a packet arrives at one port, it is copied to the other ports so that all segments of the LAN can see all packets.
HTTP	Hyper Text Transfer Protocol - data transmission for application layer, e.g., websites.
IGMP	Internet Group Management Protocol (IGMP) is a communications protocol used by hosts to report their multicast group memberships to IPv4 routers.
IP	Internet Protocol – used to build logical units (subnets) in a network.
Latency	Delay introduced by packetizing or buffering - number of samples per frame divided by sample rate – also known as 'frame size'.
Managed Switch	A managed switch can be configured to prioritize LAN traffic so the most important information gets through. An unmanaged switch on the other hand behaves like a "plug and play" device. It cannot be configured and simply allows the devices to communicate with one another.
mDNS	Multicast DNS – resolves host names to IP addresses, part of zero conf.
Multicast	One sender to many receivers.
Packet	Formatted unit of data – consists of control information and user data (payload).
Packet Time	The real-time duration of the media data contained in a media packet. For example, a packet containing 24 samples of 48 kHz audio has a packet time of 24 ÷ 48 kHz = 500 microseconds. Short packet times allow for lower latency but introduce overhead and high packet rates that may overtax some devices or networks. Long packet times imply higher latency and require additional buffering which may not be available on memory - constrained devices.
(PoE)	Power over Ethernet. This allows a single ethernet cable to provide both a data connection and electric power to devices such as wireless access points, IP cameras, and VoIP phones.
PTP	Precision Time Protocol - used to synchronize clocks throughout a network-defined in IEEE 1588-2008.
QoS	Quality of Service - overall performance of a network. QoS is a feature of routers and switches which prioritizes traffic so that more important traffic can pass first. The result is a performance improvement for critical network Technical Guide | HelixNet IP Network Guidance Updated Apr 2021 Page 21 traffic. QoS equipment is useful with VoIP phones or in LANs with high volumes of local traffic.
Router	A router is a device that joins networks together and routes traffic between them. A router will have at least two network cards (NICs), one physically connected to one network and the other physically connected to another network.
RTP	Real Time Transport Protocol - used for transmission of real time data.
RTCP	Real Time Control Protocol - controls quality of transmission and negotiates QoS parameters.
RTSP	Real Time Streaming Protocol - controls media streaming server.
STP	Spanning Tree Protocol is a network protocol that builds a logical loop-free topology for Ethernet networks. The basic function of STP is to prevent bridge loops and the broadcast radiation that results from them.
Static IP	A static IP address is an IP address that was manually configured for a device, versus one that was assigned via a DHCP server. It's called static because it doesn't change.
Subnet	A subnetwork or subnet is a logical subdivision of an IP network. The practice of dividing a network into two or more networks is called subnetting.
SDP	Session Description Protocol- describes the configuration of a stream.
Session	Describes the stream parameters (audio format, number of channels.
TCP	TCP, (Transmission Control Protocol) is a standard that defines how to establish and maintain a network conversation via which application programs can exchange data. TCP works with the Internet Protocol (IP), which defines how computers send packets of data to each other.

UDP	(User Datagram Protocol) is an alternative communications protocol to Transmission Control Protocol (TCP) used primarily for establishing low-latency and loss tolerating connections between applications on the Internet.
Unicast	Point to point connection between sender and receiver.
URL	Uniform Resource Locator- references to a resource on a network.

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 or 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.

Alameda, California, USA
Headquarters
Tel: +1.510.337.6600
Email: info@clearcom.com

Carlsbad, California, USA
Operations & Manufacturing
Tel: +1.858.535.6000

www.clearcom.com