VALCOM

A (P) POTTER COMPANY

Best Practices and General Troubleshooting Procedures

Revision 2023-1.26

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Engineered, Built and Supported in the U.S.A.

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"Installing the best equipment using known best practices does not cost, it pays"

When qualified integrators follow best practices, most installations of Valcom equipment go flawlessly. However, occasionally things don't go according to plan. After reviewing this document and employing some of the troubleshooting techniques, if you have not reached a satisfactory conclusion to the issue at hand, then give us a call at 1-540-563-2000[1].

This document is an ongoing effort with contributions from many. Content is added and edited quite often.

Technology is always evolving. We learn, and share, new troubleshooting techniques all the time.

Are you new to public address system design? Start here.

We strongly encourage you to check for document updates often by following the update link found in the footer.

This PDF contains embedded hyperlinks to additional detailed information, click on hyperlinked text or graphics to view this information.

Best Viewed With

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[1] Technical support staffed M – F 8:30 a.m. to 7:30 p.m. EST (Roanoke, VA U.S.A.)

Contents

Recent Change Log

2/3/23 – Added Packet Sender to Useful Software Utilities section

6/5/22 – Added Audio Fun Facts

3/5/22 – Added a note about VIP-102B scanning issues with USB/Ethernet adapters

1/29/22 – Added info on troubleshooting hosted SIP firewall issues

12/1/21 – Revised Cannot Access individual multizone intercom/PA channels and Reboot Troubles or Reboot Required Troubles

11/10/21 – Added "If noisy power supplies are suspected of inducing noise into a system, temporarily substitute the supplies with DC batteries (2 9-volt batteries connected in series will power most 24-volt speakers and circuits). If the batteries quiet the noise, then the power supplies are the likely noise source".

10/30/21 – Expanded upon Managing Multi-facility VoIP Installations

10/26/21 – Added A partial list of reasons why a network endpoint won't communicate

9/23/21 – Added to VIP-102B Scanning Issues - Red "error status" ICONs indicate a routing issue between the computer hosting the VIP-102B and the endpoint(s). They may also be the result of invalid DNS entries having been assigned to the endpoint(s) and/or a time out issue requiring adjustment in the VIP-102B to facilitate waiting longer for endpoints to respond.

9/15/21 – Added info to SIP (Session Intiation Protocol) Troubles

10/1/20 – Added Testing Talkback Switching

9/25/20 – Added Redundant Designs

9/17/20 – Added The Blessing of Battery Backup

8/31/20 – Added Troubles Following a Power Failure

7/31/20 – Added Upgrading Firmware

Technology Choices

Before we begin, a little history from our [Design Guide](#),

Before integrated circuit technology, most public address speaker systems used old fashioned 100, 70.7 or 25-volt central amplifiers and heavy gauge shielded wire. While old fashioned central amplifiers are still used today, a more common approach allows the use of small gauge UTP and self-amplified or IP based technology.

Self-amplified or IP based technologies allow for easy installation, functional versatility, large scale deployment, and remote maintenance. [UTP](#) based systems are always the best choice for large, busy facilities since they inherently support many simultaneous announcements and/or talkback intercom conversations.

Old fashioned centrally amplified systems are restricted to one broadcast per central amplifier at any given time.

Self-amplified speakers feature a non-blocking, independently volume controlled, matched amplifier per speaker. They connect to the audio source and a shared dc power supply using UTP cabling.

IP speakers communicate over a properly configured network and offer all the benefits of self-amplified technology plus more.

IP speakers connect and are powered via PoE network switch ports (one per IP speaker), are independently addressable and feature virtual software-based volume adjustment and programming. IP speaker systems easily span multiple buildings, cities, states and countries.

Talkback, or 2-way intercom capability, is an option with any style of speaker or horn. Speakers or horns used as talkback intercom points are typically IP, 45-ohm or old fashioned 25-volt speakers and are often accompanied by a separately mounted pushbutton (call button). The button allows users to ring a telephone. Once the telephone is answered, a bidirectional connection is established with the talkback intercom point.

45-ohm talkback speakers may or may not have built-in attenuation. It's important to choose models that are compatible with your intercom head end equipment. There is no difference between the 25-volt speakers used for talkback or one-way audio. Their functionality is determined by the system to which they are connected.

Talkback is an automatically switched, hands-free connection and does not require push-to-talk. The talkback conversation is terminated when the telephone terminates the call.

Talkback intercom points are quite common in classrooms, hospital operating rooms, elevators, building entrances, medical examination rooms and car dealership mechanic bays. Talkback systems should not be expanded to more than 2 talkback speakers per output, and because each speaker acts as a microphone, perform best in relatively quiet environments. Having multiple speakers picking up background noise and speech can make it difficult to discern the audio from the person with whom you are conversing.

Full IP Intercom

Valcom IP Mass Notification/Intercom systems differ from analog wired systems in numerous ways.

Hosting facility systems on a LAN/WAN provides many benefits, not the least of which - long term cost savings. Managing multiple IP based systems typically requires fewer personnel since most adjustments and diagnostics may be performed remotely. That means less time lost driving to sites, fewer service vehicles required, less vehicle insurance cost, less fuel cost, and more multitasking.

Unlike analog systems, Valcom IP Mass Notification/Intercom systems do not require a central control system. They are hosted on the LAN/WAN, therefore the physical location of endpoints and their proximity to each other is irrelevant. Also, unlike analog systems, system size constraints are essentially non-existent. These systems are easily deployed on a facility, enterprise and/or global scale.

Valcom's server-less design means that if properly configured network connectivity exists between endpoints, they will be able to communicate. This robust, redundant strategy coupled with inherent supervision, explains why Valcom IP Mass Notification/Intercom systems are utilized in some of the most vital facilities in the world.

A full complement of one way or intercom PoE speakers and horns are available to suit any area. These speakers and horns *connect as network endpoints* and may be selected in any combination conceivable for announcements to a single area, multiple areas, or everywhere. Access may be via single line POTS type telephones, FXO ports, loop start C.O. line ports, loop start trunk ports, SIP, microphone or analog station ports (FXS) featuring CPD (Calling Party Disconnect).

Visual notification devices, such as LED displays, may easily be incorporated into your design to deliver messages to high noise areas, to benefit hearing impaired individuals, or anywhere that visual alerting is desired.

Multichannel Talkback Gateways are available in 45-ohm or 25-volt models. They may be used to upgrade existing analog intercoms. Each channel provides an adjustable talkback audio feed to either 45-ohm or old fashioned 25-volt intercom speakers. All models offer one or more line level outputs for direct connection to an amplifier line level input or to self-amplified speakers. Additionally, one normally open call button input is available per channel.

Input/output Gateways allow users to launch messages from panic buttons or automatically from 3rd party monitoring devices. They also provide switch outputs to control electric door locks, lighting or any other facility system.

Audio gateways allow the introduction of music, microphone or other external audio sources. They may also provide audio outputs to facilitate integration of existing legacy analog paging systems, radio systems, etc.

Although the Valcom IP Mass Notification/Intercom systems feature a server-less design, there are Application servers available to provide desirable features.

Telephone Paging Servers allow the broadcast of system announcements through the speakers of many existing IP telephones. This simple addition adds audio coverage to private offices and other areas that may not be close to a system speaker.

Application Servers may be added to provide scheduled tones, music, prerecorded or live announcements. They also provide a graphical browser interface to launch messages or monitor call status. Application servers feature the ability to monitor data feeds such as syslog, RSS, ATOM or CAP feeds to automatically launch one or more messages to speakers/horns, IP telephone speakers, text to LED signs, as well as screen pop ups on PCs. All the messaging modes may be simultaneously initiated from a single user action.

Advanced Servers allow users to launch their own emergency announcements via CAP, RSS and/or ATOM feeds. This allows the incorporation of any system capable of responding to such feeds.

All servers have a high availability option.

What's the best solution?

There is no overall best solution. It depends on the requirements of any given site. Both self-amplified and IP systems are widely deployed and used as a standard worldwide.

While full IP deployment offers the most versatility, it's not uncommon to use IP gateways and Application Servers for audio distribution between local wiring closets while using analog self-amplified or talkback speakers and horns as the actual sound points.

This is all part of the many benefits of Valcom's technologies. There are typically multiple options to achieve the end user's internal communication goals.

The choice of using VoIP or analog equipment for the main control and audio distribution is dependent upon many factors:

a) The preference of the owner
b) The intent of the system (live audio announcements, music, etc.)
c) The size of the system
d) The number of sites included in the system
    a. If properly configured network connectivity exists between the Valcom IP gateways and endpoints, they will work together. That's a very strong advantage of choosing an IP solution; you can have as many "main" and "remote" sites as are necessary. Systems installed in different geographical areas may all act as one. Unless a single site is providing specific services, losing network connectivity to one site does not affect the others at all. For this reason, IP is the best choice for large installations.
e) The feature set required
f) Initial budget
g) Desire for long term cost savings
h) Expertise of the installer
i) Expertise of the owner
    a. Systems that will be maintained by a facility's IT department are typically full IP deployments.
    b. Systems maintained by facilities management are typically full IP, analog or a hybrid of both.

We will gladly assist you in choosing one or more options for your application.

You only get paid for a job once

Several years ago, a school district released an RFP for 4 of their schools. The RFP was to replace the clock and intercom systems over the Summer break. As it turns out, the jobs, which were virtually identical in scope (same equipment, same size, etc.) were awarded to 2 different local integrators.

The 2 jobs that were awarded to Integrator A were completed in a month or so and experienced no issues whatsoever.

The 2 jobs that were installed by Integrator B, however, had a myriad of issues and were not even completed when school reopened.

The equipment was the same, the environment was the same, so why the difference? The difference was that Integrator A took the time up front to do things correctly.

That experience prompted the start of our best practices checklist, which has now been incorporated into this document.

Since that time, we've seen many changes in the industry. What was once primarily analog in design is now network and microprocessor based. Because of this, we've seen many new additions to our integrators' field support, all well practiced in networking and programming.

Unfortunately, many of the basic troubleshooting skills, a large percentage of which were acquired from working in the analog realm, have not yet been acquired by the newest generation of field technicians.

The purpose of this document is:

a) To share the best practices we've learned over many years from thousands of integrators, best practices that will save you time and money.
b) To teach and preserve the basic troubleshooting techniques that are as important today as they ever were.

A Successful Job Starts at The Site Survey….

Whenever possible, conduct an on-site survey of the intended jobsite. An ideal time to visit is when the site is in full operation so that you can gather accurate noise level and activity data from the site.

Site surveys can provide vital details that RFPs alone cannot. RFPs don't typically advise of costly accessibility challenges like asbestos laden ceilings, heavy room furnishings that must be moved to complete work, requiring a rented scissor lift to mount horns, or concrete walls through which infrastructure must be routed.

We've prepared a [Site Survey Form](#) to assist you in your site survey. It is available in both Microsoft Word and PDF formats.

After you conduct a site survey, an important part of a project's initial bid process is for you and your client to formally document a detailed scope of work. Unstated customer expectations can quickly lead to dissatisfaction and expensive cost overruns. Detail the customer's expectations and where your responsibility begins and ends.

This is especially important when the proper operation of the system that you are providing is dependent on systems that you do not manage, such as telephone systems and facility networks. If necessary, your pre negotiated responsibility should be limited to proving that the equipment you've provided is fully functional and that factors beyond your scope of work are preventing successful deployment. You might even contractually agree upon steps to determine this.

Remember, while you are installing, the customer is thinking. Failure to define a scope of work upfront will result is the customer's expectations growing in proportion to install time.

Do not commit to reusing any existing infrastructure or equipment without a caveat of "if compatible and in good condition". Agree to a process of change orders should the customer expand their expectations or if site conditions are not as described in the scope and warrant additional charges.

Site access is closely related to defining the scope of work. Agree upon times when the contracted work can be performed. Oftentimes, the best time to work is after a site's regular business hours, however, work related areas may be locked at these times. Pre-negotiate unrestricted access to all areas involved in the scope. Having your technicians wait for a security guard or custodian to open a locked room is never a profitable situation.

Best Practices

"Best practices are the result of experience. Experience? Well, that's often the result of bad practices"

Experienced integrators know that the best way to avoid or minimize troubleshooting time and effort is to take the time to neatly organize, label and thoroughly document the system during installation. Any existing equipment that will be integrated with the new system (amplifiers, phone system ports, computer networks, wiring, speakers, microphones, etc.) should be independently assessed and those assessment results should be documented before any new equipment is connected. If multiple technicians are involved in the installation, adhering to a common game plan will result in consistency of techniques and quality. Taking the right steps up front can be the difference between a profitable and non-profitable job.

It's much easier to troubleshoot a system when:

a) Those involved understand the intended operation of the system.
b) Up to date "As-Built" documentation is available.
c) All cables are labeled and neatly routed.
d) Unique cable colors and consistent wire pair color codes have been used.
e) All infrastructure terminations are labeled.

f) All equipment is visibly labeled with identifying information (areas served, MAC address, IP addresses, etc.).
g) New and existing subsystems have been independently assessed and the results have been documented.
h) Backups of any programming have been archived and cataloged.
i) A detailed summary of system operation is available.
j) A system of cross-connect blocks and patch cables is employed, allowing technicians to easily make and change direct connections between two different termination blocks (one for field wiring and one for equipment) within a termination closet.

■ **Read all the equipment manuals. Countless troubleshooting hours may be saved.**

■ Consider pre-staging head end equipment. Many integrators will pre-stage and pre-program an installation before moving the equipment to the jobsite. This practice offers many benefits. Pre-staging allows integrators to address any issues in the resource rich and comfortable environment of their facility. Once satisfied that "all systems are go", the on-site install should proceed without incident. Should new troubles occur on site, they will typically be diagnosed as environmentally introduced. This is the best method of impressing your customers with a seamless, trouble-free installation.

■ Never assume that your test equipment, like meters and lineman's handsets, are working properly. If a test yields unexpected results, then use your test instrument on a known working device to verify that it is working properly.

■ To prevent unintentional connections and for potential future use, always *individually* tape and store unterminated conductors.

■ The twist in UTP wiring serves 2 purposes. One is to promote equal noise coupling into each conductor of a pair so that the noise is cancelled by a differential input. The other purpose is to cancel the inherent electromagnetic field that surrounds ac carrying conductors – the differential signals on the conductors are 180 degrees out of phase with each other, therefore the EMF from the conductors are also 180 degrees out of phase. The out of phase EMFs cancel. For these reasons, the integrity of the twist in UTP should be maintained right up to the termination points. In addition, when combining UTP

Single Conductor   Parallel Conductors   Twisted Pair

Emission   Emission   Emission
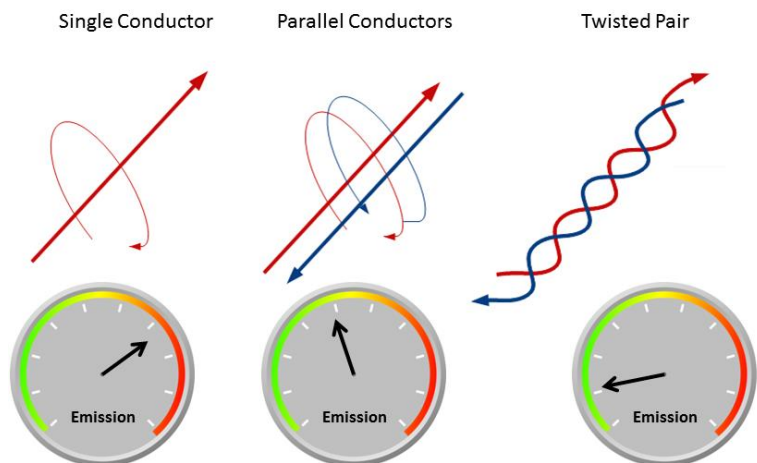
pairs to increase AWG, all "tip" sides and all "ring" sides should be connected (i.e. W/B & W/O as one conductor and B/W & O/W as a second conductor) as opposed to using whole twisted pairs as a conductor for one side of the signal.

■ To minimize noise, loss and RFI issues, keep all wire runs as short as possible. Locate amplifiers, power supplies, network switches, etc. as close to the endpoints they serve as practical. Remember that the audio source and the power supplies for Valcom Self Amplified speakers and horns are commonly located in different areas. A single power supply, unlike an old-fashioned central amplifier, may power speakers and horns in multiple zones.

■ To help prevent induced noise, do not mount audio equipment or audio equipment power supplies near power transformers, even if they are on the other side of a wall.

■ To help prevent induced noise, do not mount audio equipment or audio equipment power supplies near high power radio transmitters.

■ To help prevent induced noise, do not route audio cables near florescent light ballasts.

■ To help prevent induced audio coupling, do not coil extra cable lengths.

■ To prevent coupling between audio inputs and outputs, always route the 25-pair cable connecting to the Class Connection or MultiPath CPU card separate from all other cables.

■ *Refrain* from fixing things that <u>are not</u> broken. For example, do not upgrade firmware just because an update is available. Many techs have inadvertently introduced troubles in the process.

■ Manufacturers may, or may not, advise planning for ventilation. If uncertain, don't overcrowd equipment racks or backboards. Allow for convective heat transfer from the equipment by providing room for air to circulate. Excessive heat typically shortens the life of electronic equipment.

■ Plan for dedicated, non-switched, 15-amp ac outlets wherever your equipment requires ac power. Many troubles have been the result of sharing ac circuits with inductive loads and/or people inadvertently switching outlet power off.

■ Be aware that Ethernet based equipment may have specific network requirements.

■ To minimize wiring cost and facilitate troubleshooting, it's best to locate PoE switches and power supplies in IDF closets within each speaker area. For self-amplified speakers, use cross-connect blocks to connect distributed audio and locally supplied dc voltage to individual speaker cables.

■ 110 type blocks used for audio connections save space but may make troubleshooting difficult. It's much easier to access connections on 66 type blocks.

■ Do not attempt to terminate stranded or heavy gauge wire on 66 or 110 [blocks].The pins are not designed to pierce heavy insulation or to make a good connection to stranded conductors.

■ To help prevent equipment (and human) damage, always remove power from equipment before service.

■ Standardize on a wire pair color code (w/bl pair for audio; w/br pair for call switches, etc.). Standardization eliminates many errors and facilitates troubleshooting.

■ Use uniquely colored cables and patch cords for associated cabling. For example, purple cable for clocks, yellow cable for dc voltage, blue cable for audio. Doing so will save time in identifying cables both during installation and for future maintenance. There are many cable colors available. Many of these have use dictated by ANSI/TIA/EIA-606, so coordination with the facility network administrator will be necessary.

■ Allow non-PoE endpoints an opportunity to negotiate network communications by making network connections before applying power.

■ Consider future maintenance when planning equipment placement and cabling. Always leave enough extra cable to account for adds/move and changes. Plan to facilitate changes. Leave room to work!

■ To prevent equipment damage in Valcom analog intercom systems, never mix powered and un-powered system cards on the same ribbon cable.

■ When installing wireless clocks, always install and power the transmitter first so that wireless signal will be available as clocks are installed.

■ When connecting audio from one system to another, take time to make sure that the audio levels and output vs. input impedances are compatible. For efficient signal transfer, a source's output impedance should be <= the load's input impedance. For maximum power transfer a source's output impedance should be equal the load's input impedance.

■ When planning a site installation, Give some thought to connectivity. For example, if a device providing dial tone has RJ11 jacks and is connecting to another device that connects to dial tone via a 66 or 110 type punch down block, you will need to have RJ11 cords, RJ11 wall jacks and cross-connect wire to make connections between the two.

■ Use cross-connect blocks and patch cables for MDF and IDF connections in order to facilitate troubleshooting, adds, moves and changes. Terminate all cable pairs on the 66 or 110 blocks for future use.

Speaker and Call Switch Cabling – Punch down all pairs for future use

Equipment Cabling

Interconnect Cabling
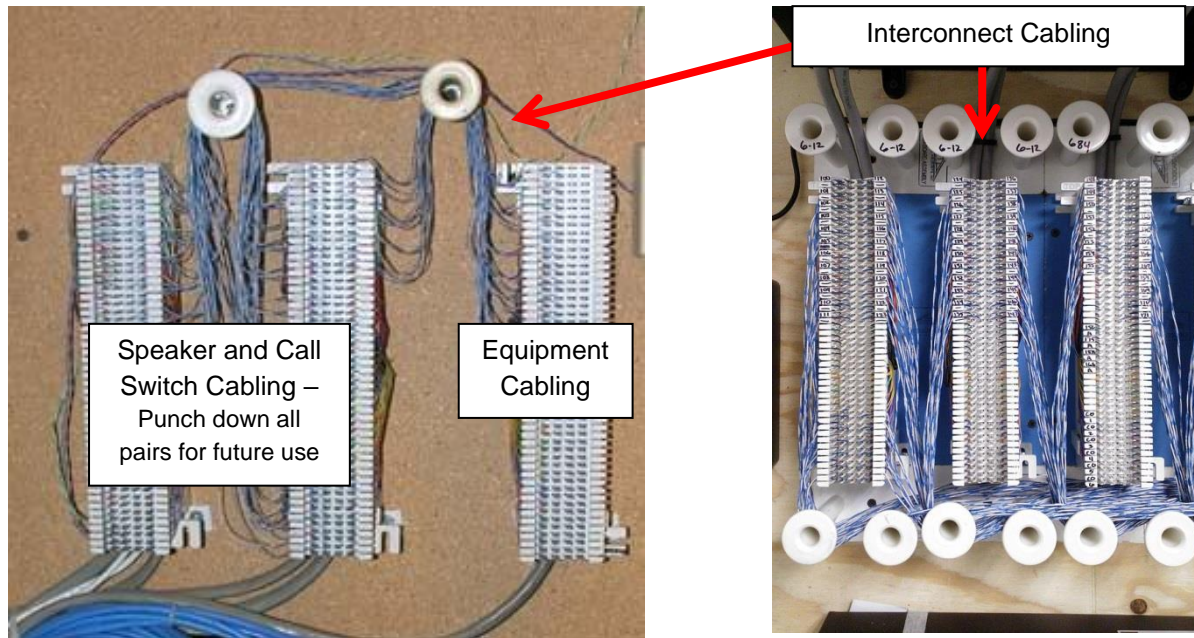
■ Always double check card addresses before powering up a Class Connection or MultiPath system. Having multiple system cards with the same address can cause damage.

■ To prevent noise and intermittent operation, power supply wire and audio connections should be tight and secure. Poor wiring techniques will almost always result in system noise and other troubles.

■ When starting up VECPU5 systems with RIDFs, use the system programming software to connect to the CPU and view the system configuration screen to see that all cards are present. This quickly verifies VERCA card operation.

■ When connecting line level audio from one system to another, consider using VMT-2 Audio Isolation Transformers to avoid ground loops (which result in noise or worse)

■ To allow future volume adjustment, V-LPT attenuators should initially have their volume control set to at least halfway.

■ If your design will include analog speakers or horns mounted in locations that will be difficult to access after installation, or includes areas with many speakers connected in parallel, add convenient wall mount volume controls to make future adjustment easier.

■ When installing volume controls, adjust the speakers or horns for maximum required volume with the volume control turned all the way up. This will allow the most volume adjustment range for the controlled speakers or horns. When call switches with volume controls are utilized in classrooms and other areas, the maximum speaker volume will be determined by any volume settings/connections made at the actual speaker. Setting talkback speakers to their maximum tap setting will provide for full range from the call switch volume control.

■ Save yourself time and frustration by calling the factory for help before spending hours troubleshooting. If you're not comfortable with the answers you get, ask to speak to someone else.

■ Even if your cellphone has good reception onsite, plan for a wired telephone jack near your installed equipment locations. Factory support often requires a dependable, conveniently located, telephone connection.

■ Computers that are used for programming systems should have Internet access. Factory support often requires remote access via the Internet.

■ Loud areas are not well suited for talkback communications; therefore, 25-Volt speaker circuits used for 2-way talkback are usually designed for a speaker load of 1 watt or less.

25/70.7/100-volt speaker circuits measuring higher wattages should be connected to an appropriate amplifier output for one-way broadcasts.

■ Always pre-check speaker lines, new or existing, for transient voltage, short circuits to ground and proper impedance before interconnecting to the intercom equipment. Use a good quality **Impedance Meter**, not an Ohm meter.

Speaker Type	Number of speakers	Sum of Tap settings	Expected Impedance Reading
45 Ohm	1	N/A	=> 45 Ohms
45 Ohm	2	N/A	=> 22.5 Ohms
25 Volt	Any	<= 1 Watt	=> 625 Ohms
25 Volt	Any	<= 5 Watt	=> 125 Ohms

■ An accessible label on a VoIP device indicating its MAC address can save a lot of time. Once a VoIP device is installed, the MAC address may be inaccessible.

■ When installing IP based intercom components, it is crucial to accurately record each device's MAC address versus installed location.

■ Although the practice of using spare conductors in talkback speaker cables to carry 2-wire digital clock correction signal is advisable (in MultiPath and Class Connection ES Systems only) and can represent a significant wiring cost savings, using spare conductors in <u>one-way</u> speaker cables to carry 2-wire digital clock correction signal will result in undesirable noise over the speakers. Do not do it!

■ To prevent crosstalk, telephone circuits should never be run in the same cable as speaker circuits.

■ If a connection doesn't seem right, or an assembly does not seem to fit together correctly, call the manufacturer to confirm compatibility before continuing.

■ When installing equipment, wall speakers and surface mount clocks for example; consider future maintenance by ensuring that external screws are easily accessible. There have been many instances where such equipment is installed before the ceiling tile is in place. Once the ceiling tile is installed, it's virtually impossible to access the screws required to remove the equipment from the wall.

■ Paging speakers or horns located by telephones or microphones that are used to originate page announcements may cause feedback. There are several techniques to avoid such feedback. One technique is simply to delay any page announcements destined for broadcast through local speakers. Another technique is to create different group/all call dial codes for each telephone/microphone that excludes local speakers.

■ Modern classrooms and meeting rooms contain a myriad of electronic systems. These systems may interfere with each other. It is advisable to fully test the interoperability of all electronic systems in a model room before deploying systems facility wide. This includes smartboards, occupancy sensors, projectors, intercom, network endpoints, wireless systems and sound systems.

■ Testing self-amplified speaker lines as you install them with a tone generator is simple and provides assurance that the speaker installation is correct and functional. Speakers in common areas should all initially be set to the same volume level.

■ In the course of installing new non-Valcom network endpoints, or network maintenance, others may move Valcom VoIP devices from assigned VLAN ports thus rendering them inaccessible. The use of locking RJ45s may be used to discourage this practice.

■ Plan for granularity. When cabling speakers/horns in a facility, connecting them all on one looped cable pull (even if feasible) limits you to one zone of audio. All audio will go to all speakers due to the common cabling. Conversely, if you connect each speaker on its own cable all the way back to the main equipment, then speakers/horns may be easily

configured in any future combination desired. In most installations, a wiring plan between these 2 extremes is best. Loop speakers in common areas (multiple hallways, restrooms, etc.) that will always receive common general announcements. Use dedicated cables for speakers in areas that will likely require area specific audio or omission from general announcements (classrooms, boardrooms, each floor, lobby, etc.)

■ Consider wiring speakers in common areas such that adjacent speakers are sourced from different wiring closets, amplifiers, power supplies and audio outputs to insure the continuation of audio coverage should one system component fail. The speakers can still work as one zone by creating groups.

■ Witness reported troubles firsthand. It is quite common for trouble reports to be inaccurate or incomplete. For example, you may receive a report that users are unable to make announcements in a certain building, where in reality; they are dialing an incorrect access code that will not allow them to make announcements anywhere.

■ When reusing existing site equipment, such as speakers, call switches and wiring, that equipment should be tested and recertified before connecting to the new Valcom equipment. Sometimes existing speakers are tapped improperly or have cracked cones, existing wiring may be shorted to ground or have an inadequate number of conductors for the new system, etc.

■ Unless advised to do so by a manufacturer representative, never upgrade firmware in a device that is functioning as desired.

■ If it doesn't fit, don't force it – find out why it doesn't fit.

■ Master keys to a site should never leave your possession. We know of an integrator that lost a master key to a site – it cost him $30,000 to have the site rekeyed.

■ Pretest any equipment that would be time consuming to reinstall.

■ Pre-install the latest versions of all required software programming and diagnostic software prior to visiting a jobsite. Software install issues caused by protection software or lack of administrative rights can cause extensive site work delays.

■ As is true of many things, audio amplifiers, speakers and horns are not typically intended to operate at full rated capacity/volume 100% of the time. A system design requiring audio devices to continuously operate at maximum volume is a poor design and much more likely to experience audio quality troubles and premature equipment failure.

■ System ports that provide dial tone **must not** be directly connected to any other ports that simultaneously provide dial tone.

■ Unless directed to do so by a manufacturer representative, system outputs that provide voltage **must not** be connected in parallel to any other outputs that simultaneously provide voltage.

■ System outputs that provide audio **must not** be directly connected to any other outputs that simultaneously provide audio.

■ To prevent noise and undesirable circuit loops, transformer isolate system audio inputs before connecting them to a common audio source. Valcom's VMT-2 1:1 Isolation Transformers may be used in this capacity. Simply connect each system audio input to the audio output of a dedicated VMT-2, and then connect the common audio source to the audio inputs of all the VMT-2s.

■ Default Valcom VoIP devices before moving them from one system to another, Valcom VoIP devices retain their programming. Moving them from one system to another without defaulting will pollute the new system with undesired audio groups and dial codes.

■ Jobsites in areas designated as having historic or cultural significance may be protected by laws or rules that regulate the types of improvements that may be made. Be aware of any such regulations before making a proposal.

■ Horns and speakers have a rated frequency response. Applying audio signal frequencies to a speaker or horn that are outside of the usable frequency range may have detrimental effects. As an example, loud paging horns cannot typically produce low frequency audio. Sourcing low frequencies to these horns can damage the horn coil due to the misuse of the applied energy (heat). This is the reason why some amplifiers have optional high pass filters to roll off low frequencies.

■ If a piece of equipment fails, take time to investigate why it failed. We've seen too many instances where damaged equipment is replaced without determining the cause of the damage, only to have the replacement fail again. Field damage is not covered by warranty.

■ Make it a habit to backup all programming (VIP-102B Snapshots, all system server(s) programming, other system programming) whenever you begin work on an existing system *and* whenever you complete programming or maintenance of a system. Always save all backups with names identifying what they are and the date and time they were saved. Include a text document with the backups describing the changes that were implemented. Keep copies of backups both on site, and off site.

■ Most relay outputs on Valcom equipment are light duty "control" relays. These relays are not designed to directly switch power to anything requiring more than 1 amp of current and 30vdc open circuit voltage. When controlling more than the rated voltage and current,

the relays on Valcom equipment should be isolated from the actual load with an appropriately sized heavy duty contactor (slave relay). The slave relay should be selected such that the input coil's voltage and current requirements are within the specifications of the Valcom control relay. The output contact(s) of the selected slave relay should be rated for at least twice the voltage and current requirements of the anticipated load.
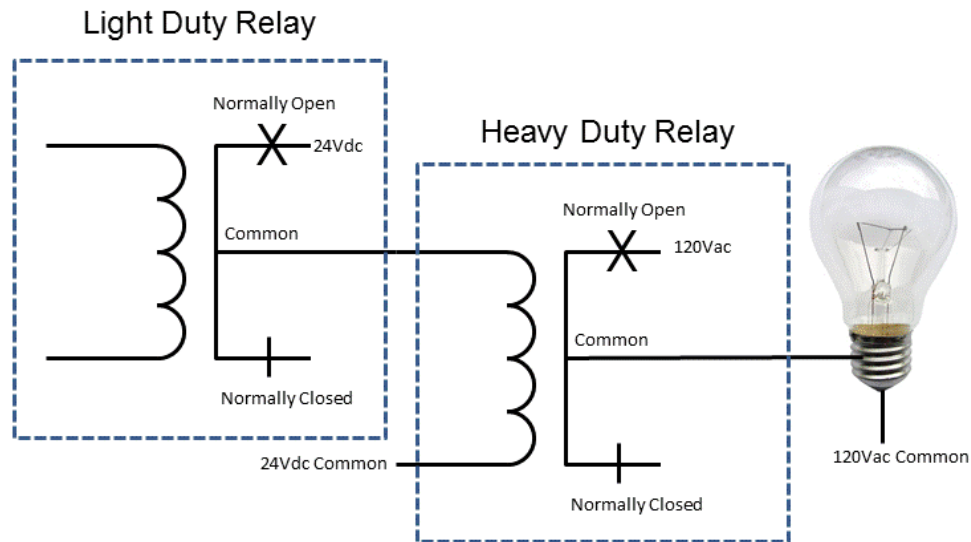
■ To prevent damage from lightning EMFs, any audio system cables routed exterior to a facility, especially aerially routed cables to outside horns or between buildings, should have appropriately sized surge protection on both ends of the cable. If the installation site is located in an area prone to lightning damage, then consideration should be given to adding appropriately sized surge protectors to all system I/Os.

■ Surge protectors are available for many different needs and it's important to choose models that are appropriate for your application. Improperly sized surge protectors have been known to restrict signal magnitude resulting in clipped audio signal delivery to speakers or horns. This clipped signal results in low and or/distorted sound.

■ Be aware of, and comply with, all environmental requirements for the equipment involved in your installation. Most manufacturers publish an acceptable operating temperature range and tolerable humidity specifications for their products.

■ Visualize a site installation before submitting a proposal. It's very easy to forget expensive resources such as a rented lift for mounting horns to high ceilings and ac outlet installation in equipment closets.

■ When multiple system components have time and date settings, it's important to use a common time base (like NTP) so that they are all set to the same time, date and time

zone. Not doing so can cause some system functions, like processing CAP messages to fail.

■ It's a good practice to use the same polarity audio for all amplifiers and all speakers in an area. This is accomplished by wiring them all the same way:

a) If multiple amplifiers are serving the same area and share a common audio source then connect them all to the audio source output in exactly the same way.

b) Always connect the inputs of neighboring speakers (marked + on some speakers) with the same exact polarity.

Speaker cones push out and pull in as the applied signal voltage changes from positive (+) to negative (-). This movement pushes and pulls the air and causes vibrations that we interpret as sound.

If all speakers are in phase, then they are all pushing or pulling the air at the same time. Since the pressure waves are all going in the same direction, the sound pressures combine and reinforce each other.

If neighboring speakers are out of phase with each other, either because of their wiring, or because they are connected to out of phase amplifiers, then their sound pressures negate each other.

In Phase Audio   **Out of Phase Audio**

■ Talkback systems should not be expanded to more than 2 talkback speakers per output, and because each speaker acts as a microphone, perform best in relatively quiet environments. Having multiple speakers picking up background noise and speech can make it difficult to discern the audio from the person with whom you are conversing.

■ The speaker to phone audio quality of 2-way talkback speaker locations directly in the path of active fans or HVAC vents will be noisy at best. Choose a better location.

■ Most music sources have left and right channel stereo outputs. Stereo audio, by definition, cannot be supported by public address systems. Therefore, the source's left and right channels must be mixed to form a single monophonic output. It's not unusual to visit sites where only one of the stereo channels is connected to the PA system or where both stereo channels are connected to one music input. Either will result in poor music quality and possible damage to the audio source.

Valcom's Remote Input Module (V-9130-W) provides an inexpensive means to properly mix stereo audio to monophonic audio.

■ IP systems that produce high quality foreground music use a significant amount of bandwidth and require a network capable of providing that bandwidth. Valcom IP systems are optimized for voice and should not be promoted for high fidelity music distribution.

Assume Nothing.

Question Everything.

VIP-102B IP Solutions Setup Tool Best Practices

The VIP-102B IP Solutions Setup Tool is a Windows based application that is used to program Valcom VE and VIP VoIP devices. This section contains some Best Practices pertaining to programming Valcom IP systems with the VIP-102B IP Solutions Setup Tool.

This is not intended as a tutorial. For that purpose, a video example may be found here and a reference manual may be found here.

■ *Always make certain that you are using the latest version of the VIP-102B Software Tool.*

■ Read the IP6000 Initial Setup Procedure or Valcom VoIP Network Requirements document. These documents contain a lot of good advice on initial setup and choosing dial code lengths. The IP6000 Initial Setup Procedure is for Valcom Engineered Solutions projects (has part numbers that begin with VE), the Valcom VoIP Network Requirements is for standard Valcom VoIP designs (no part numbers begin with VE).

■ Often networks hosting new Valcom VoIP installations will require additional setup during system programming. Arrange to have the facility Network Administrator readily available to troubleshoot/modify the network as required.

The biggest problem we encounter with Valcom VoIP cutovers is with integrators not reading the aforementioned documents, and therefore, not having the network properly configured.

VIP-102B Tool General Best Practices

■ The VIP-102B is packed full of useful auto assign and copy functions. If you need to make the same changes to multiple devices, look for a copy button.

■ **Device/Remove All Devices** is not as scary as it sounds. It simply removes devices from the current scan. All things remaining equal, the devices will reappear in subsequent scans.

■ It's possible to make audio sent to some groups louder or quieter than audio sent to other groups. This is accomplished under **System/Volume Offsets**. The volume offset is determined by the group's priority level.

■ Individual channel volumes may be adjusted on the device **Channel** Tab(s), or global adjustments (absolute or incremental) may be set under **Programming/Volume Adjustments**. Depending upon firmware level, the latter will be a real-time change.

■ When new devices appear in the device tree, they are identified by their MAC address. Use the **Properties** Tab of each Valcom VoIP device to provide a meaningful name. This will help you identify devices going forward as the name you assign will replace the MAC address in the device tree. One can also save keystrokes on single channel devices when adding name on the **Properties** Tab, as that name will auto populate the Description field on the **Channels** Tab. Likewise, the Dial Code entered on the **Channels** Tab will auto populate the CID Number and CID Name fields. The CID number and Name fields can be manually changed without impacting the name appearing in the device tree or properties tab.

■ If you wish to change the static IP address of a <u>single</u> device, you do so manually from the network tab window. Hi-light the current address, make the change, then update the device. NOTE: make sure the address has not been assigned to another device on the network. See VIP-102B troubleshooting section.

If you wish to change the static IP addresses of <u>multiple</u> Valcom VoIP devices, then the PC hosting the VIP-102B must be set to an IP address that is not routable to the currently assigned addresses.

■ It's possible to set access passwords on Valcom VoIP devices so that programming cannot be changed. If you use this feature, carefully record the password - **there is no "backdoor" password.**

■ Save Snapshots on a regular basis. Snapshots can even be used to copy saved programming from a previous device to a replacement device.

■ Saved Snapshots can be used to restore settings to a device if programming is accidently changed or corrupted in the future. Just open the Snapshot, perform the verification function (choosing to "Use All Tool Settings" if prompted), and update the devices to restore the previously saved settings.

■ Always save a Snapshot before programming firmware in case the firmware changes settings.

■ Valcom VoIP systems can be preprogrammed using sample devices and saved as a Snapshot. Then using "Replace Device" one can install the programming into devices on the network while on-site. Simply load the Snapshot into the VIP-102B, Choose Device/Replace Devices and click Find Devices to scan the actual network and find the real devices. Then drag the real devices to the preprogrammed Snapshot devices. Follow the prompts to continue. If you encounter a prompt for updating firmware, and you do not wish to do so, simply continue to the next screen. As a note, Find Devices will only find devices with different MAC addresses from those in the Snapshot. ***If necessary, prior to replacing a device's programming, update the Snapshot so that the network tab(s)***

of the device(s) shown will be routable to the programming PC. Replaced devices will not properly update if restored data includes non-routable network settings.

■ All saved Snapshots should have a unique name identifier (not just "Snapshot") and its name appended with a date to easily determine the latest Snapshot. The name identifier should in some way identify the scope of the Snapshot. Subsequent Snapshots of the same scope should use the same name identifier with different appended dates. Subsequent saved Snapshots on the same date can use A, B, C, etc. or -1, -2, -3 to additionally append the date.

 Examples:

All PBSC_012716

PBSC North Campus_012816

■ Most Valcom VoIP devices can have both a channel dial code and a SIP identity. These can be the same number. The SIP identity is used to access the device from a SIP enabled telephone system. The channel dial code is used for everything else.

■ You don't have to update after each change; however, your changes are not saved until you do.

■ If you work with a lot of different systems, you might want to periodically do some housekeeping with the **File/Clear File Cache** option to remove all the locally cached INI files. These INI files can grow quite large and slow down the VIP-102B.

■ Take a moment to examine the **File/Options** menu. There are several options to control how the tool works, especially to turn off some prompts and confirmation messages if you find them annoying. If you turn off the conflict prompts, you can still know if there are conflicts present because the button on the toolbar with the yellow exclamation point will be enabled to warn you.

■ Program all of the Valcom VoIP devices to use the same NTP (Network Time Protocol) server (**Network** Tab) and the same **Hours Offset from UTC** (**Properties** Tab). Certain system functions may fail if there is a time and/or date discrepancy between devices. For example, CAP messages stamped with the wrong date/time may appear as expired to recipient systems.
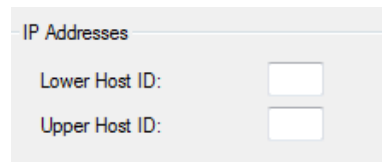
■ Most PCs can print in the form of a [PDF](). Print a PDF of system programming (File/Print Reports). This PDF can be converted to an Excel spreadsheet using a converter (such as the one found at https://www.pdftoexcelonline.com). This allows easy editing for providing end user dialing plans.

■ The VIP-102B can be used to upgrade firmware in Valcom VoIP devices. Firmware should never be upgraded unless suggested by Valcom. If you do upgrade firmware,

initially upgrade one device and then continue with a few devices at a time carefully verifying that the upgraded devices are performing as expected after each step.

VIP-102B Tool Network Scanning & Setup Best Practices

■ When defining Subnets, only enter IP address upper and Lower Host IDs that you will actually be using (plus 10% or so for expansion). Don't enter all addresses from 1 to 254 in a subnet unless they are going to be utilized. The VIP-102B will attempt to contact each defined IP address during scans. Scanning unused addresses unnecessarily slows the scan process.

IP Addresses

Lower Host ID:

Upper Host ID:

■ If there are no domain names to resolve in your system, then do not populate the DNS server entries when you define VIP-102B network subnets.

■ Valcom VoIP devices and PCs running the VIP-102B all produce both a multicast beacon and a broadcast beacon, which allows users to ascertain minimal routabilty to system components. This is not a thorough GO/NO GO test but is a good indicator of basic network configuration. These report in **Communications/Network Diagnostics**.

■ If you need to identify an endpoint after installation, and do not know the MAC address, open Communication/Network Diagnostics in the VIP-102B. If the VLAN and PC are properly configured, all endpoints will report in via a broadcast and multicast beacon. Unplug the Ethernet connection of the device in question and Network Diagnostics will no longer receive its beacons.

■ A successful scan, where all devices are discovered, does not necessarily mean that the VLAN hosting the Valcom devices is properly configured. When scanning, the VIP-102B uses several methods to discover devices, it scans by Multicast, it scans the local area network, it scans any additionally defined networks and it waits for beacons from VIP devices that did not previous respond.

■ When scanning a brand-new system, you can expect dial code and IP address conflicts as all of the Valcom VoIP devices will have factory default programming.

■ When adding devices to a system, (new or previously used), always check the "Use Defaults" checkbox.

■ When adding new endpoints to an existing system, ensure that they are connected to the VLAN, and or, are routable to the other Valcom endpoints. Scan to discover the new

endpoint(s), assign the desired IP address, rescan, check "Use Defaults" for the new devices and program as desired. Be certain to rectify any conflicts before updating the system.

■ Initial VIP-102B scans report in a window that allows you to filter the results by all or part of the device names or IP addresses. When working on networks that host multiple Valcom IP systems, use filters to bring in the devices from one system at a time.
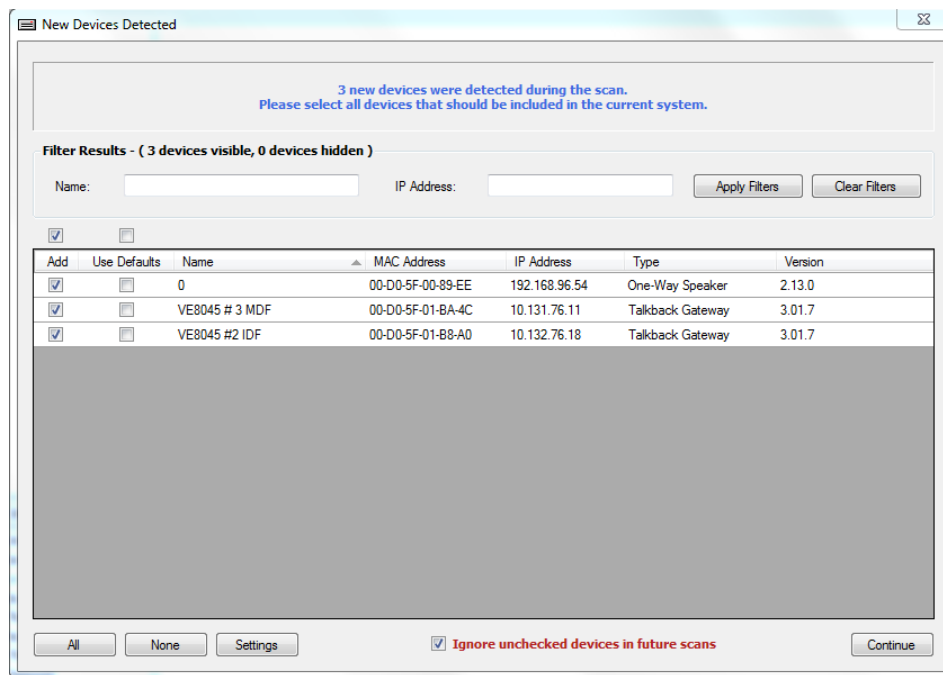
■ You'll notice an option to Ignore Devices – it appears several places in the software. If you ignore a device, then it will not scan into the VIP-102B until it's removed from the Ignore List (**Device/Ignore list**).

■ There's a recycle bin icon in the VIP-102B. In systems with an Application Server, if a previously scanned system device is no longer available, then the VIP-102B will show this device in the "recycle bin". If the device is truly removed from service, then remove it from the "recycle bin" or it may create conflicts because its IP address and dial code are reserved in case the device reappears (just in case part of the network is temporarily off line).

■ If there is a problem communicating with a Device, right click on it and View Cached Files. There are several log files for each device that are often helpful in diagnosing problems.

VIP-102B Tool Dial Codes and Conflicts Best Practices

■ As stated before, when scanning a brand-new system, you can expect dial code and IP address conflicts as all the Valcom VoIP devices will have factory default programming.

■ Eliminate the dial code conflicts by assigning a range of temporary dial codes (**Programming/Assign Dial Codes**). Initially choose a range of dial codes that will not conflict with the actual dialing plan for the facility. For example, if you plan to use 3-digit dial codes from 100 to 400, initially set all the dial codes from 600 – 900.

■ Otherwise, when removing dial code conflicts, remove invalid length codes first, clear the recycle bin as needed, rebuild the phone book, and then tackle individual conflicting codes.

■ The VIP-102B automatically builds a "phone book". Occasionally you'll encounter phone book conflicts. You can manually rebuild the phone book under **System/Phone Book**.

■ When managing multiple systems that are routable to each other, if you want common groups between systems, like "all call" to all systems, independently add the same group code to each. Failure to adhere to this will result in all dial codes in all routable systems polluting all other systems. It basically makes programming more difficult due to unnecessary clutter.

■ Clear all conflicts before updating devices.

■ All VIP-102B assigned dial codes must be the same length. Dial Code translation allows you to translate between shorter dial codes and longer dial codes. For example, if your system uses 5-digit dialing, but you want to use zero for all call, Dial Code translation will access the all call group whenever zero is the first digit dialed. Be aware, if you use dial code translation, no other dial codes can begin with the digit(s) assigned to be translated.

■ If you have defined one or more networks/ranges of static IP addresses (**PC ICON** in Upper left corner), then it's easy to eliminate the IP address conflicts by automatically assigning IP addresses (**Programming/Assign IP addresses**) to the Valcom VoIP devices.

VIP-102B Tool Setup Best Practices

■ The PC used with the VIP-102B should be directly connected to the VLAN hosting the Valcom IP equipment. Users should make exceptions in, or disable, all port blocking firewalls and PC protection software.

■ If the Valcom VoIP devices will be hosted on their own dedicated PoE or PoE+ network switches, then use the VIP-102B IP Solutions Setup Tool to program and test the system before the switches are connected to the facility's LAN/WAN.

■ Static IP addresses are preferred over using DHCP. To assure device access, Valcom VoIP devices set for DHCP will self-assign a static IP address upon boot up if the DHCP server cannot found. This self-assigned address will be on the 192.168.6 subnet. We've experienced many instances of DHCP servers requiring 10 minutes or more to recover from facility power cycles. If DHCP is utilized, it is important that the DHCP server and all network components have battery backup.

In addition, in many systems using dial codes to activate emergency messages or emergency sequences, the IP address of the Valcom VoIP devices is used in the triggering process. If DHCP endpoints obtain a new IP address, the triggering will not work. To prevent this in DHCP systems, DHCP reservations are recommended.

If DHCP *is* utilized to automatically assign IP addresses, and the DHCP server provides DNS addresses, it must provide at least valid 2 *valid* DNS addresses. If only one DNS address is available, then the DHCP server can provide the same address twice.

■ In some installations, the equipment owner will not need to browse into the system servers or perform any maintenance (adjust channel volumes, build new groups, etc.). In this case, if the new Valcom equipment will not require access to other networks, then there's no reason to connect it to the facility LAN/WAN. Stand-alone systems often make future management easier by minimizing variables introduced by the facility's IT staff.

■ Valcom VoIP systems use Multicast for a variety of functions, the most noticeable being any audio sent to groups. If you can make announcements to individual channel dial codes, but not to groups, then your network is not properly configured for Multicast.

■ Always have a non-managed PoE or PoE+ switch available for troubleshooting network issues. Temporarily moving Valcom VoIP devices to an off-network unmanaged switch is the best way to determine if an issue is network related.

■ Factory default programming includes auto destinations on FXS Gateway channels. You'll probably want to remove the auto destination.

■ When building groups, FXS gateway channels are available to add to the groups. If you add FXS gateway channels to a group, the terminal device (phone, trunk port, etc.) will ring when that group is dialed. Once the terminal device is answered, it will join the group announcement in progress.

■ It's important to note that the control, setup, debug and audio multicast addresses for all of the Valcom servers, gateways and endpoints can be changed, however, the values for equipment intended to work as a system must be the same.

■ If you are using an audio gateway channel as an input, do not include the channel in any groups. An Audio Gateway channel cannot be used as an input and an output simultaneously. Audio gateway channels used as audio inputs may be controlled by an Application Server; VOX activated and directed to a specific auto destination group or switch activated and directed to a specific auto destination group. The VIP-102B setting for each follow:

Audio Activation Method	Use VOX Activation Check Box	Input
Application Server	Unchecked	Not Active
Switch on Input*	Unchecked	Audio On / Audio Off
VOX (Voice Operated Switching)*	Checked	Not Active

* An auto destination group must be defined on the Audio Gateway **Channel** Tab

■ Each I/O gateway input that will be used to trigger messages on an Application Server must be set to control the server.

■ On larger systems, don't use the "**Scan All Devices**" menu or button. Instead use the **Scan Missing** or **Scan Selected**. "**Scan All Devices**" will rescan and retrieve the data from ALL the devices. It's almost like starting the tool from scratch. If some devices weren't present, it's quicker to do Scan Missing so the data from the existing devices won't be touched. Or right click and scan individual devices if necessary.
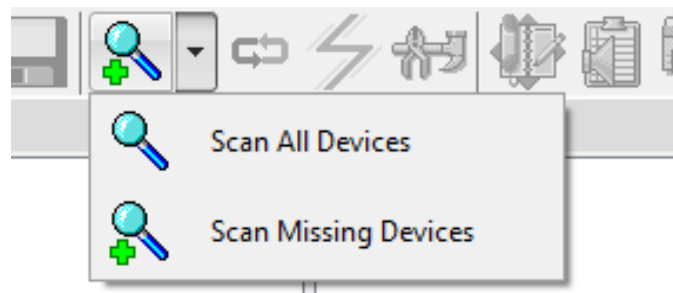
Managing Audio Groups

The VIP-102B IP Solutions Setup Tool allows the creation of multiple "audio groups". These groups are each assigned a unique dial code and are used for a myriad of alerting/messaging purposes including:

a) Making voice announcements
b) Creating Audio Events to send audio to specified speaker locations
c) Directing streaming audio to specified speaker locations
d) Creating audio recordings
e) Triggering Play Lists
f) Activating IP Strobes
g) Activating Threat Level Indicators
h) Sending Audio/Text Messages to IP Telephones
i) Sending Audio to LED Signs and PC Screens

A highly-recommended method of group management is to create unique groups for each group purpose to be performed in specific site areas. This method allows for easy management of voice, text and visual indicators on a per area basis.

By creating individual "function per area" groups, and logically labeling them for easy identification, users can easily manage alerting modes:

DIAL CODE	GROUP NAME	PRIORITY	PURPOSE
000	General All Call	40	General Announcements
001	Emergency All Call	50	Emergency Announcements
002	General Bell Everywhere	50	Scheduled Audio Tones
003	Music	10	Streaming Music Distribution
004	1st Floor Text to LED Signs	40	Manage 1st Floor Text to LED Signs
005	Lobby Threat Level Indicators	50	Manage Lobby TL Indicators
006	Basement Strobes	90	Manage Basement Strobes
007	1st Floor Strobes	90	Manage 1st Floor Strobes

Example:

A typical facility will have a group that includes all speaker locations. This is commonly referred to as "all call". This solitary group could technically be utilized to distribute automated announcements, audio files, general announcements, emergency announcements, scheduled audio events and music.

However, by only having one "all call" group for *all* purposes, there is no flexibility to modify the audio destination for any *one* purpose.

Referring to the table above, the group membership of group 002 "General Bell Everywhere" could easily be modified to exclude certain speaker locations from receiving scheduled tones during standardized testing. Likewise, the group membership of group 003 "Music" could easily be modified to control where music is being distributed.

Being dedicated groups, modifying these 2 groups will have no effect on other group code-initiated functions.

Managing Audio Groups for Schedules

For scheduled audio, distribution management is accomplished by creating a solitary group *for every unique combination of speaker locations that will receive simultaneous scheduled audio.*

Groups are unrestricted and speaker zones/stations may be members or any or all groups.

You might have 3 separate announcement audio groups for the 6th, 7th and 8th grade classrooms; if there will be scheduled events that will distribute to all 3 grade levels, these multiple groups may be assigned as simultaneous destinations.

Name these "bell groups" with a unique prefix, perhaps "B" for bells and "A" for audio announcements, so that they are easily identified when creating schedules.

Dial Code	Group Name	Priority
020	A Everywhere	40
026	A 6th Grade	40
027	A 7th Grade	40
028	A 8th Grade	40
021	A 6th, 7th, 8th	40
010	B Everywhere	50
011	B 6th, 7th, 8th	50

Following these guidelines will simplify schedule creation and management by facilitating ad hoc modification to single schedule event groups without sacrificing the functionality of groups used for other audio distribution, such as live announcements.

Managing Multi-Facility VoIP Installations

It's not uncommon to have Valcom IP systems deployed in multiple facilities that:

a) Primarily act as separate systems
b) May be managed by one individual or group of individuals
c) Share common groups for multi-facility or enterprise-wide announcements

A common example is a school district with Valcom IP systems deployed in each school and a desire for district wide announcements. There are important considerations related to such applications.

First, determine a logical dialing plan, noting that all dial codes in the enterprise-wide system must be unique and the same length (up to 11 digits long).

When multiple facilities have common room numbers, then one or more leading digits should be used to identify the building. For example, Room 202 in the first building would be assigned dial code 1202, while Room 202 in the second building would be assigned dial code 2202.

For facilities with rooms identified by a letter suffix, the dialing plan should include enough digits to allow for this letter suffix.

For example, if a facility has 3-digit room numbers that include letter suffixes such as 202A, 202B then a trailing digit indicative of the "A" or "B" will be necessary. In building 1, room 202 would be assigned dial code 12020, room 202A would be assigned 12021 and room 202B would be assigned 12022.

Oftentimes, multiple facilities will already share a phone system and have unique dial codes assigned to the phone in each area. Using this same code for the area speaker(s) may prove to be convenient for everyone.

Remember to reserve a range of dial codes for groups. Pick a range outside of the dial code plan used for individual areas or rooms. If your system includes an Application Server, remember to reserve group dial codes that will be used to dial select "Play lists", that is, will be used to manually initiate audio and other server events.

For multi facility announcements (i.e., district-wide), add the desired group code(s) for such announcements to each facility *individually*. Once proper WAN routing is in place, all members of that group code will receive the global group audio.

Since these systems will be routable to each other, to facilitate centralized management and enterprise-wide announcements (EWAs), effectively managing the enterprise with the VIP-102B requires the use of one or more advanced techniques.

There is no reason to manage the entire enterprise for every programming change. The EWAs simply require that properly configured multicast routing exist between all Valcom endpoints (a.k.a. Devices) and that each "separate" system include one or more identical group codes (same dial code, same priority and same description) in order to process the EWAs.

Making changes to one site may be facilitated several ways.

For new installations, it's easiest to initially set up each facility while they are disconnected from the WAN, then use <u>one or more</u> of the following 3 management techniques.

1) **Technique 1**. This technique should *always* be employed. Set up each facility on a unique subnet and/or name all facility endpoints (on the Properties Tabs of the VIP-102B) beginning with a unique identifier, like SHS for Smith High School). Initial VIP-102B scans report in a window that allows you to [filter](#) the scan results by all, or the leading portion, of the endpoint names or IP addresses. When working on networks that host multiple Valcom IP systems, you may use filters to retrieve the endpoints from one system at a time.
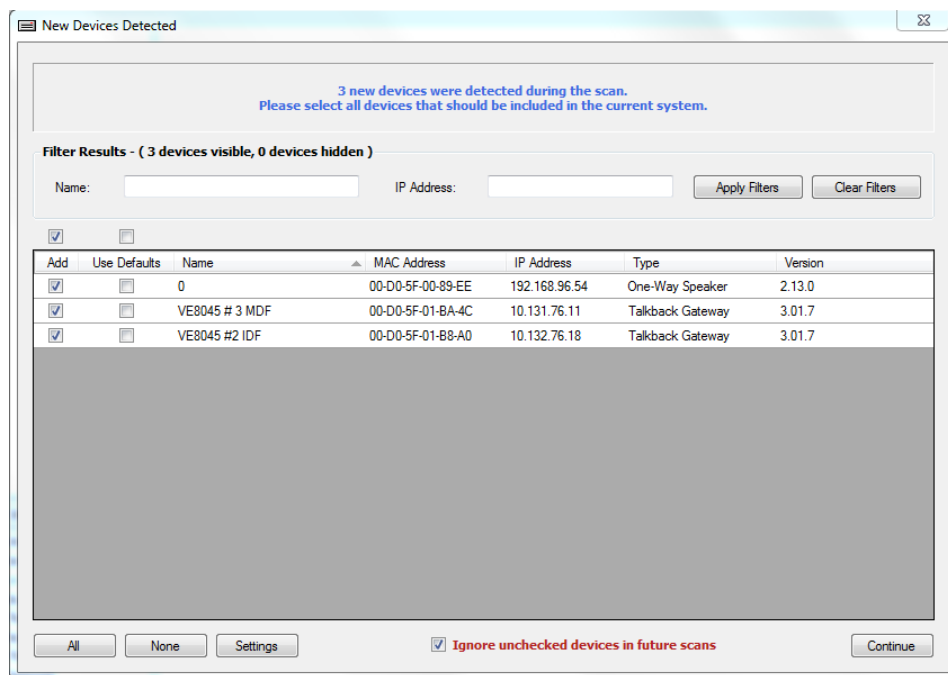
2) **Technique 2**. In most cases, this is the *least* desirable technique. Set a unique password on all Valcom endpoints in each facility (Security/Change Device passwords) Be careful to record the password as it will be required for all future programming and cannot be reset to default if forgotten. If you scan the whole enterprise, only enter the password for the endpoints in the facility currently under maintenance.

3) **Technique 3**. In most cases, this is the most desirable technique. After setting up each facility's system, take and save a VIP-102B Snapshot. When you want to

make changes to a facility, don't scan the network. Simply load the latest Snapshot for that facility, make your changes on the Snapshot, and update/verify.

Note that in order to work from Snapshots users must *be able to* scan the devices directly. (even though changes will not be implemented as a result of a direct scan). You will need UDP communications on the setup port 4097. And then you also need FTP (which typically uses ports 20 and 21) and Telnet (which uses port 23). If they have devices that use secure communications instead (SFTP, SCP, and SSH) those use port 22.

When working on existing enterprise installations, these site-specific Snapshots may be obtained by:
- a. Scanning the facility while the WAN data connection is removed
- b. Scanning the whole enterprise and filtering to retrieve only the site-specific endpoints and ignoring unchecked endpoints
4) When adding new or replacement endpoints to a specific facility:
- a. Temporarily remove the data link to the WAN
- b. Scan to find the facility's endpoints
- c. Check the "Use Defaults" checkbox next to the new additions
- d. Program as required
- e. Take and archive a new Snapshot to facilitate future site changes
- f. Reconnect the data link to the WAN.

Be certain to archive a new Snapshot after changes have been completed.

If you check the "Ignore unchecked devices in future scans", endpoints that are not checked will be ignored for the entire VIP-102B session.   ☑ Ignore unchecked devices in future scans

Any Snapshots that are retrieved for single site maintenance will, by default import the Ignore list that existed when the Snapshot was made. Therefore, loading a site-specific Snapshot will automatically filter scan results for site specific endpoints and any new endpoints that were never previously scanned. Remember to check "Use Defaults" for any newly added endpoints and to ignore any new offsite endpoints. Ignore lists may be cleared by the user while loading the Snapshot or under the Device menu.

As an additional important note, use a unique string of leading digits for endpoint dial codes in each facility in order to avoid dial code conflicts (i.e. 100XXX for one facility and 101XXX for another, etc.)

Note: scanning, retrieving and updating endpoints from multiple sites will cause cross contamination of dial codes and group data and will make future maintenance difficult. If endpoints are inadvertently scanned and retrieved from multiple sites, do not update. Click Device/Remove All Devices and start over.

Swapping Programming Between 2 Endpoints

Occasionally it might become necessary to swap the programming in 2 endpoints. This process will work for any IP endpoints from speakers to multi-channel endpoints.

Perform a complete scan of your system with the VIP-102B
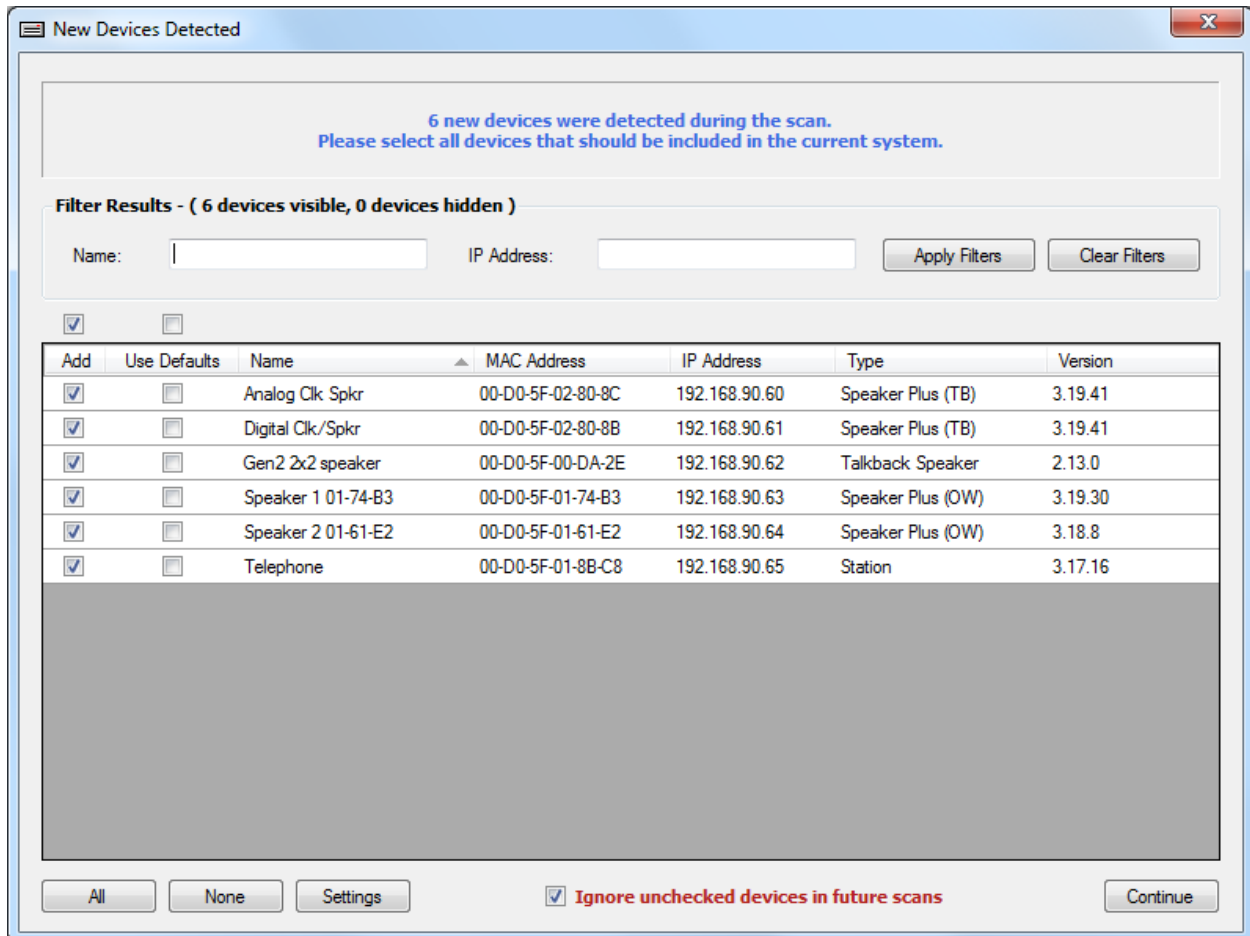
Click "Continue"

Resolve any conflicts and save a Snapshot called "Complete scan"
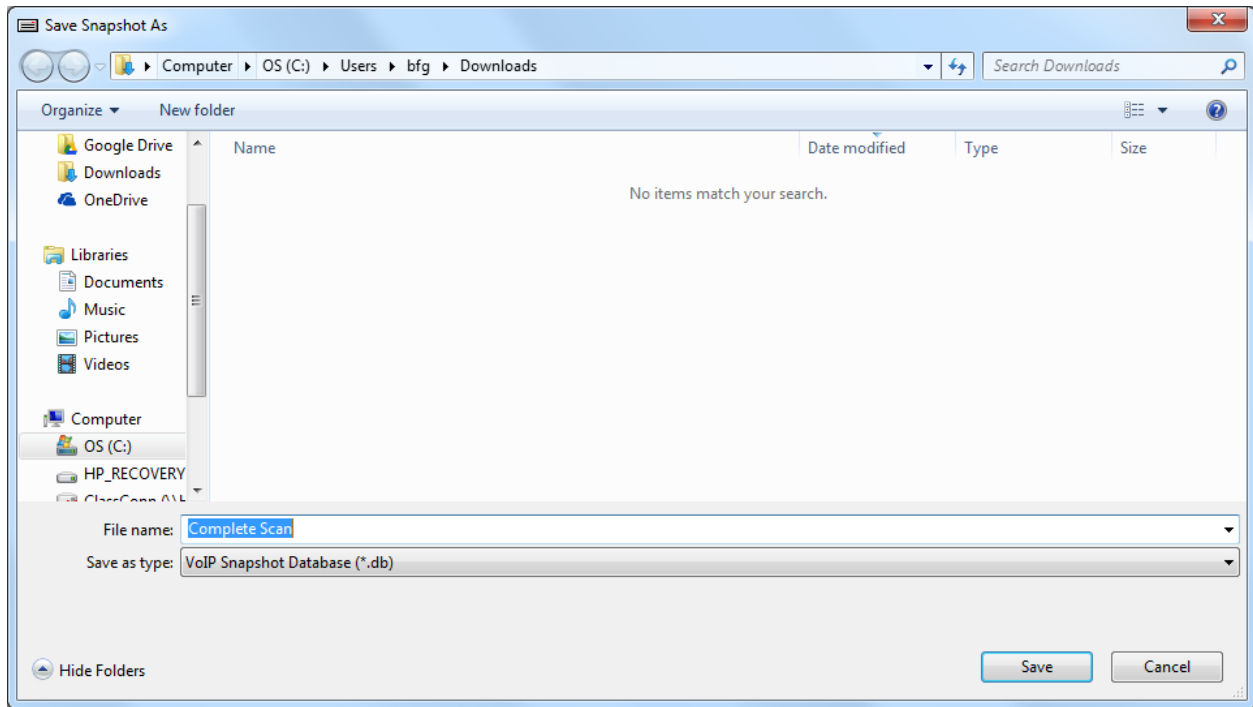
Click the "Device" menu and click "Remove All Devices"

[See it here](#)

Perform another complete scan of your system with the VIP-102B. Uncheck one of the devices that will be involved in the swap. (we'll use speaker 1) and uncheck "Ignore unchecked devices in future scans"
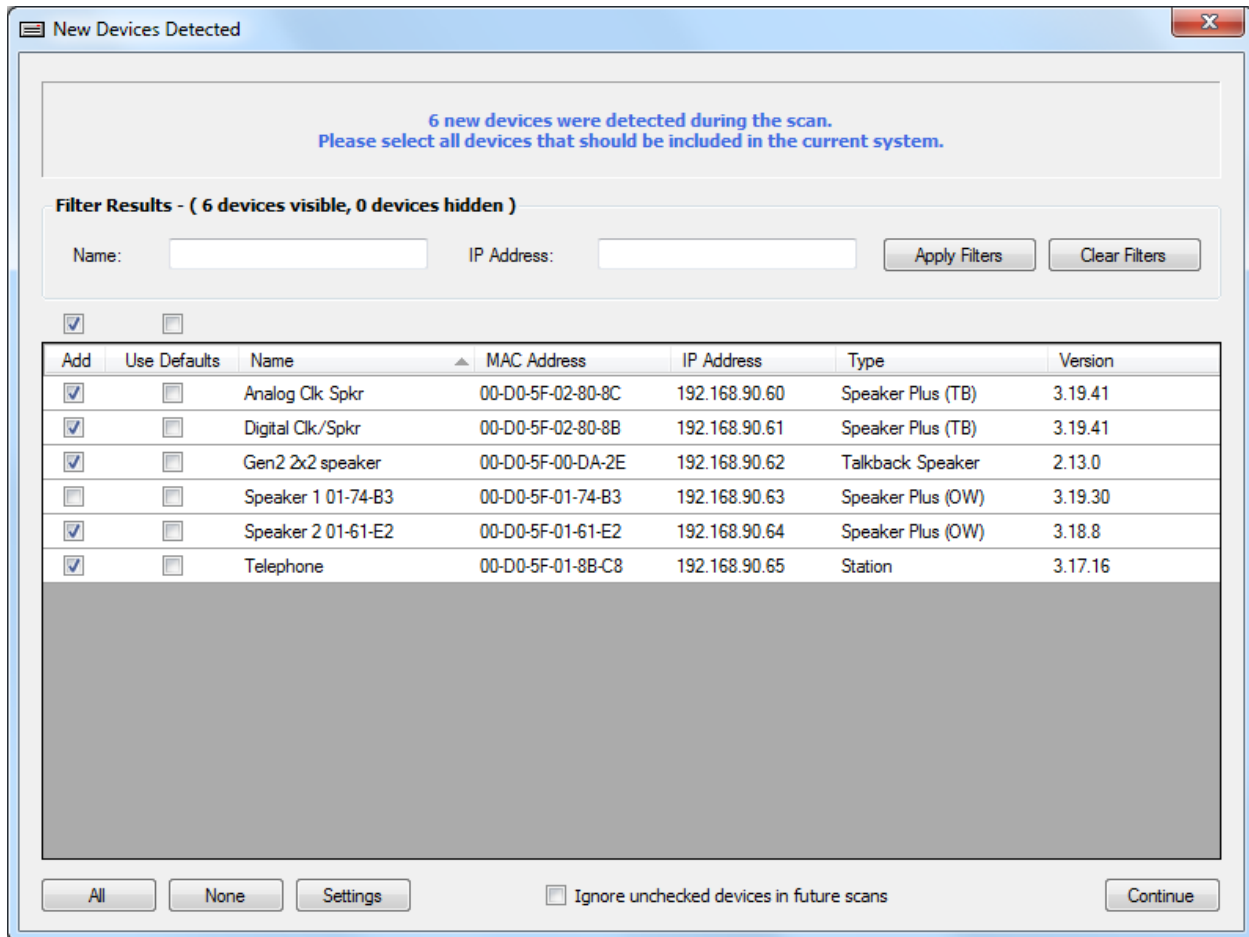
Click "Continue"

Click the "File" menu and click "Save Snapshot as . . ."

Name the Snapshot in such a way as to identify which device it omits.
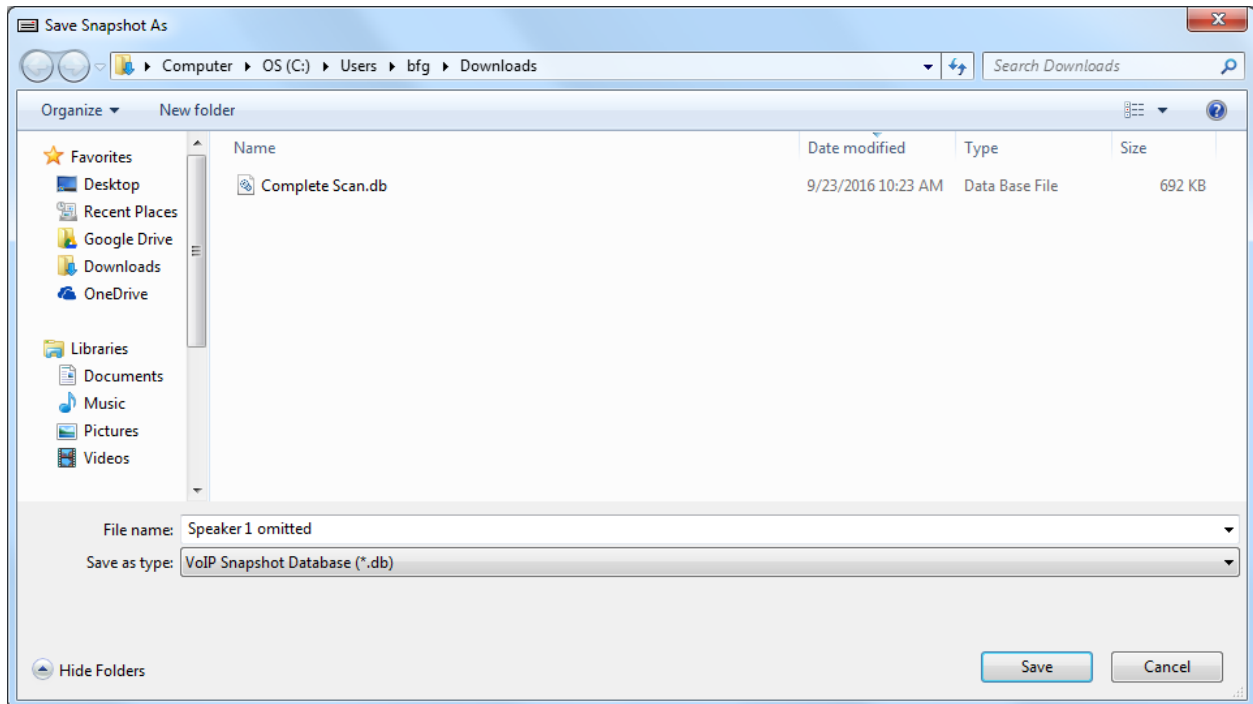
[See it here](#)

From the main screen of the VIP-102B, Click the "Device" menu and click "Remove All Devices"

Perform a complete scan of your system with the VIP-102B. Uncheck the second device (we'll refer to it as speaker 2) involved in the swap.
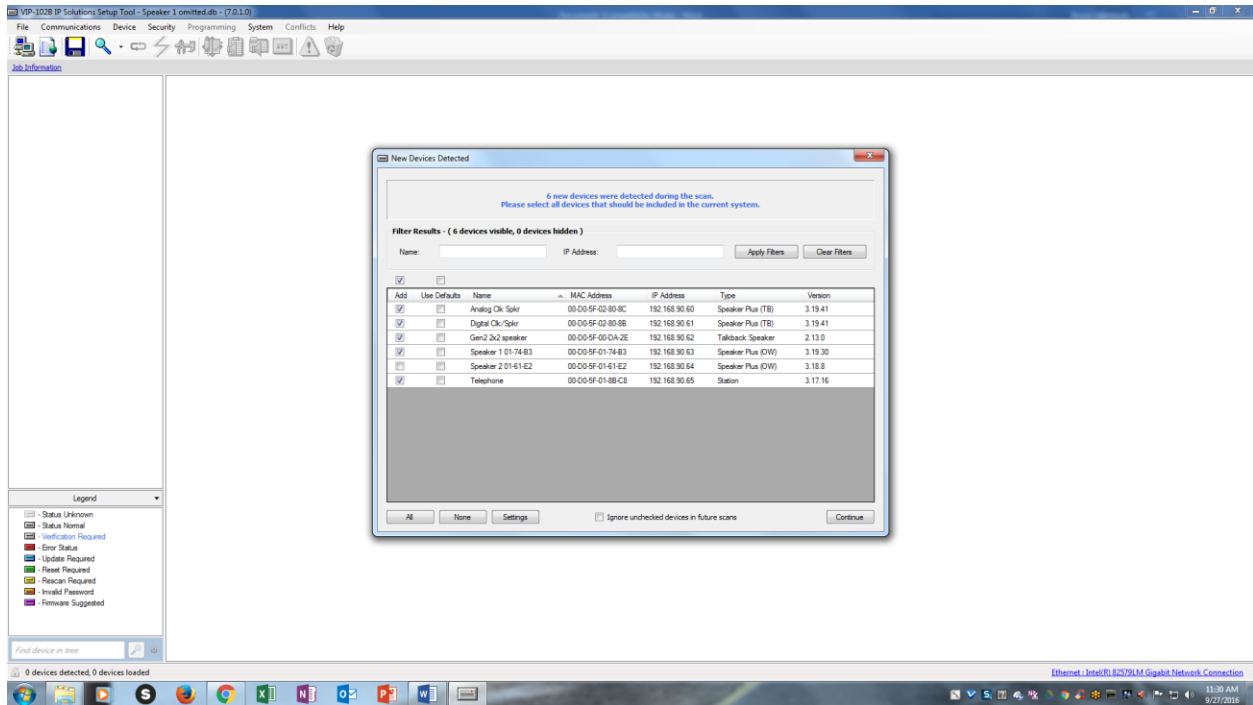
Click "Continue"

Click the "File" menu and click "Save Snapshot as . . ."

Name the Snapshot in such a way as to identify which device it omits. See it here
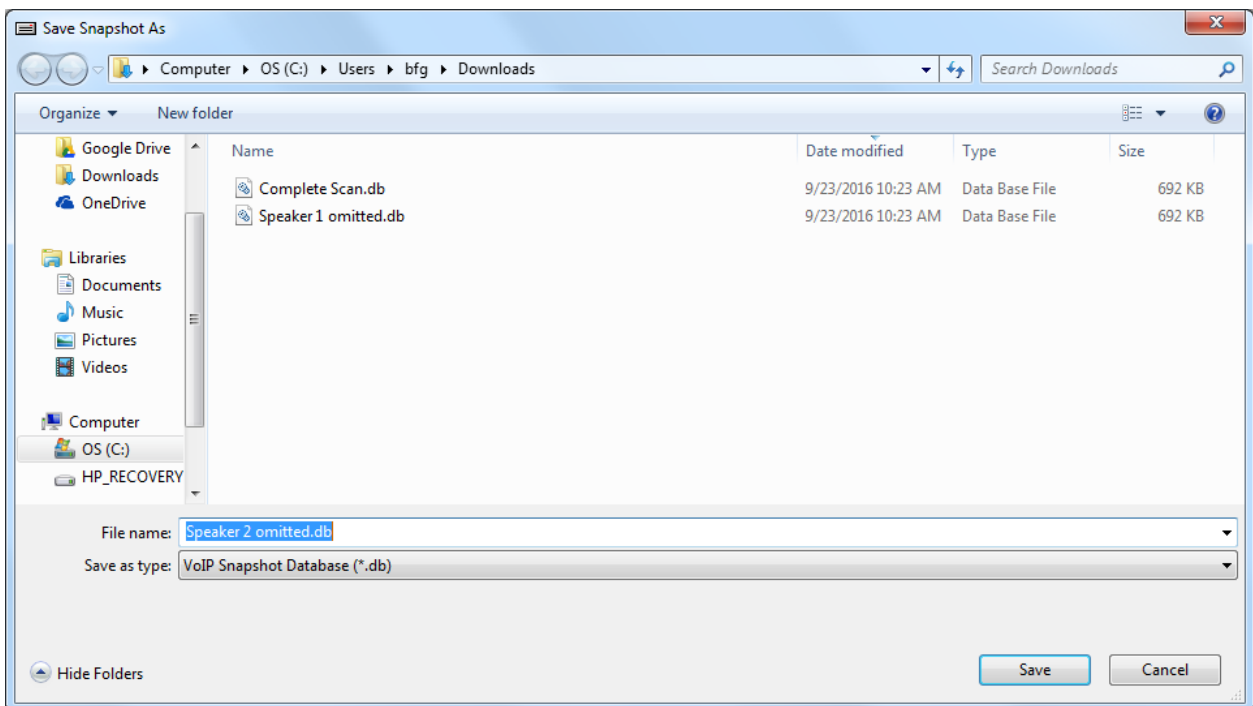
The Data Swap Process   See it here

The following steps will be repeated for each device receiving replacement programming. The following screens illustrate loading the programming from "Speaker 1 to Speaker 2".

The PC running the VIP-102B must be routable to the replacement device prior to the replacement process.

From the File Menu, open the "Speaker 2 omitted" Snapshot ", Click the "Device" menu and click "Replace Devices…"
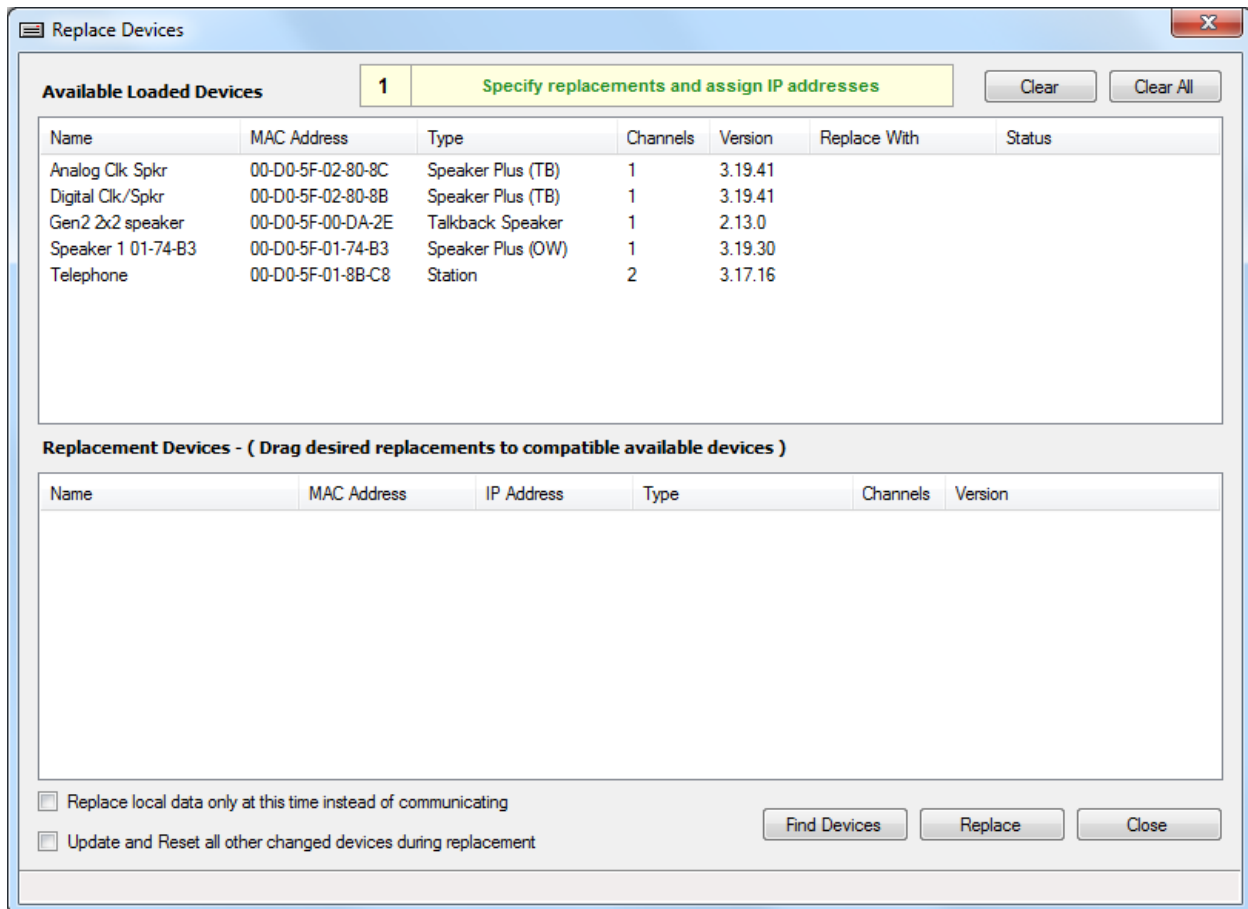
Click "Find Devices" and the VIP-102B should find the omitted device

Replace Devices

Available Loaded Devices		**1**	Specify replacements and assign IP addresses		Clear	Clear All

Name	MAC Address	Type	Channels	Version	Replace With	Status
Analog Clk Spkr	00-D0-5F-02-80-8C	Speaker Plus (TB)	1	3.19.41		
Digital Clk/Spkr	00-D0-5F-02-80-8B	Speaker Plus (TB)	1	3.19.41		
Gen2 2x2 speaker	00-D0-5F-00-DA-2E	Talkback Speaker	1	2.13.0		
Speaker 1 01-74-B3	00-D0-5F-01-74-B3	Speaker Plus (OW)	1	3.19.30		
Telephone	00-D0-5F-01-8B-C8	Station	2	3.17.16		

Replacement Devices - (Drag desired replacements to compatible available devices)

Name	MAC Address	IP Address	Type	Channels	Version
Speaker 2 01-61-E2	00-D0-5F-01-61-E2	192.168.90.64	Speaker Plus (OW)	1	3.18.8

☐ Replace local data only at this time instead of communicating

☐ Update and Reset all other changed devices during replacement

Find Devices   Replace   Close

Disconnect the Available Loaded Device (Speaker 1 in this case) from the network.

Use your mouse to drag the Replacement Device (speaker 2 in this case) to the Available Loaded Device with which you wish to transfer programming (speaker 1 in this case) and click "Replace". Follow the "Continue" prompts until the device is verified.

Replace Devices

⚠ Selected replacements will be made at this time. Please ensure that the available loaded devices that are being replaced are no longer present on the network or conflicts may prevent their replacement.

Do you want to continue?

Yes   No
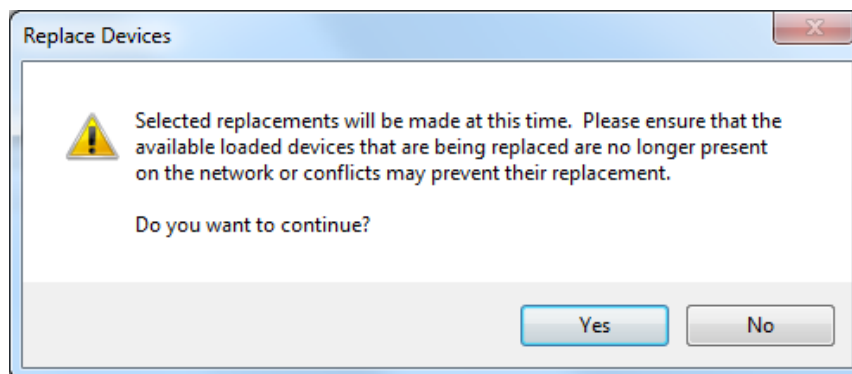
Click "Yes"

The Device will update and reset.

Click "Continue"

Distribute Replaced Devices

2	Distribute replaced devices onto the network

At this time, all replacement devices must be physically distributed and installed at their correct location on the network and the old devices that they are replacing must be unplugged and removed. Once this is complete, you may return to the tool and continue with the replacement process.

[Continue] [Cancel]

Click "Continue"

Program Firmware

3	Program firmware in devices as needed

Program By: Available Devices

Firmware File

Program By Available Devices

Check and add devices that should receive this file:

[All] [None] [Add]

Name	MAC Address	IP Address	Type	Channels	Version
☐ Speaker 1 01-74-B3	00-D0-5F-01-61-E2	192.168.90.63	Speaker Plus (OW)	1	3.18.8

* Replacement Pending

Programming List

[Change File] [Remove] [Remove All] [Program]

Device	Type	File	Status

Any required firmware programming should be done now before continuing.

[Continue] [Cancel]

Update Firmware if necessary or just Click "Continue"

Click Verify

Verify Devices

4	Verify devices to prepare for data updates

All necessary devices have been verified at this time.

Name	MAC Address	Result
☑ Speaker 1 01-74-B3	00-D0-5F-01-61-E2	Verification Complete

* Replacement Pending

[All] [None]   [Verify] [Continue] [Cancel]
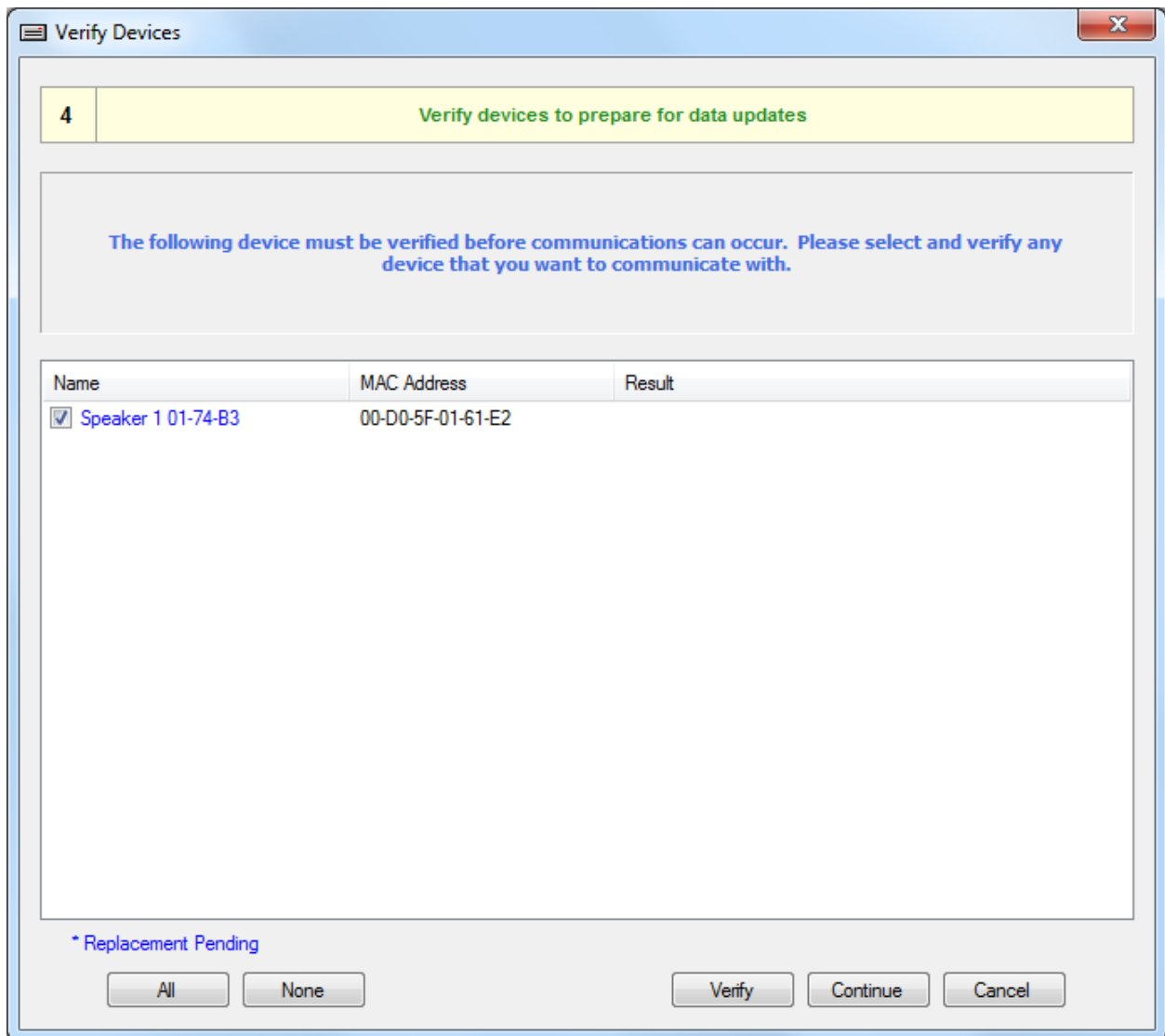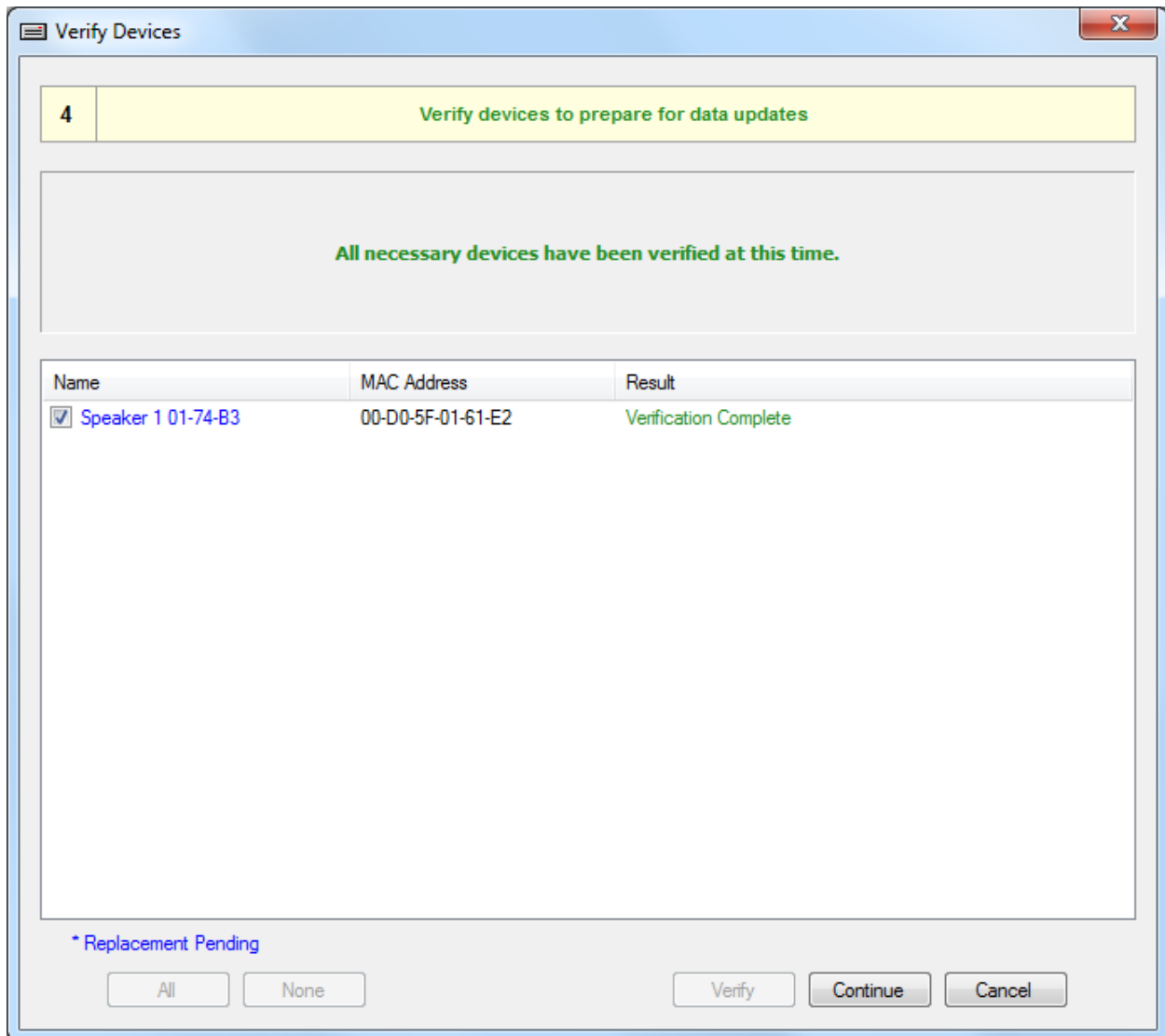
Click "Continue" then Update and Reset as necessary

Reconnect Speaker 1 to the Network.

Now start the Data Swap Process over by Clicking the "Device" menu and click "Remove All Devices",

Then choosing Open from the File menu to load the Snapshot with Speaker 1 omitted.

Once completed, click "Device" menu and click "Remove All Devices", and rescan the entire system.

The data swap will be complete.

Verify Settings Screen
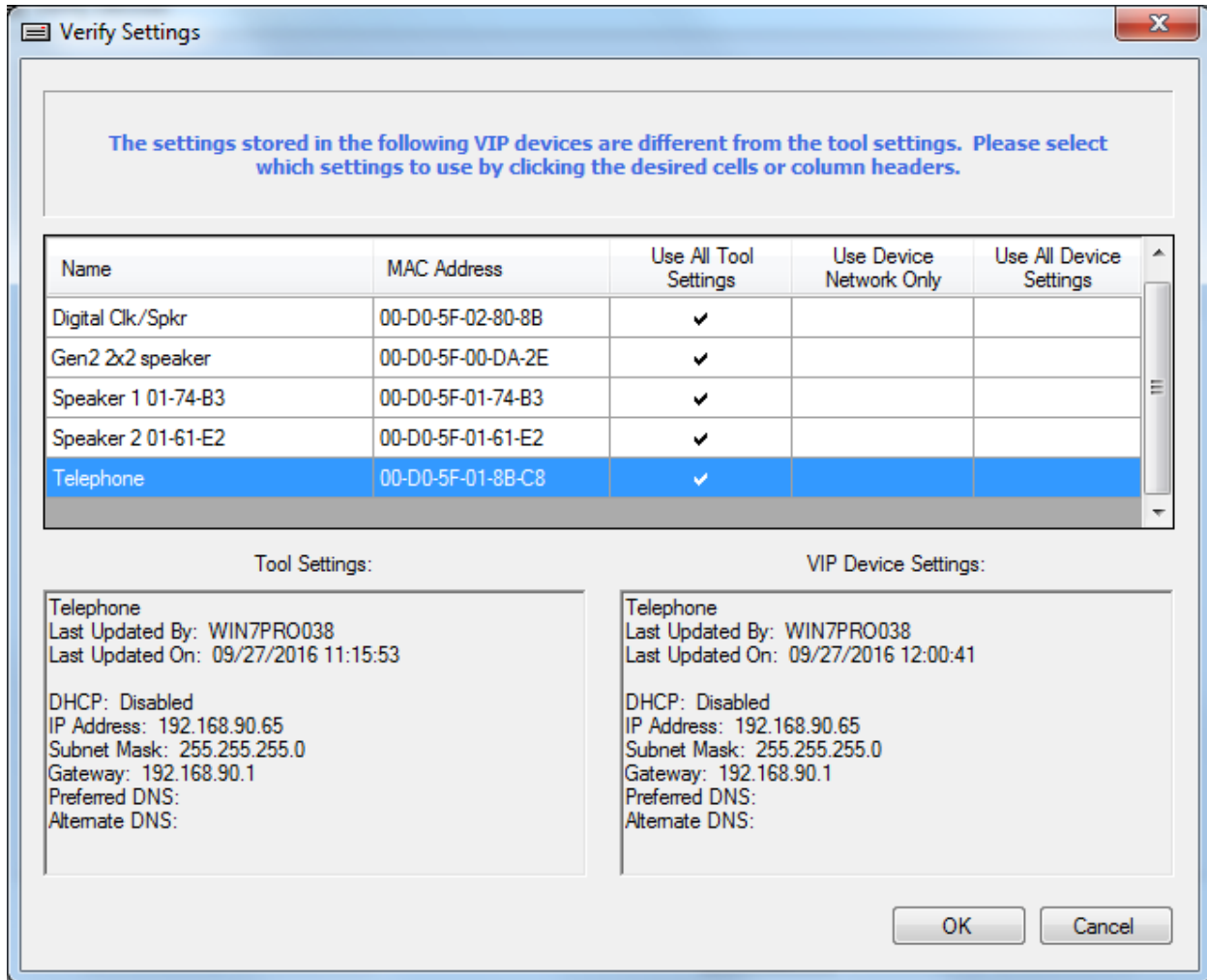
If you encounter this screen, check the appropriate boxes for which device settings to preserve.

Use All Tool Settings – use what is currently loaded into the VIP-102B

Use Device Network Only - use what is currently loaded into the VIP-102B, but preserve the device network settings

Use All Device Settings – preserve what is currently loaded into the device

Jobsite Installation Checklist

Being prepared for the jobsite is a best practice that will save time, money, and frustration. Here is a sample list of preparation questions:

First and foremost, do the installation technicians understand and have documentation supporting:

1. the scope of work?
2. intent of the system?
3. expectations of the owner?
4. agreed upon work access times?
5. expected completion and progress report timelines?
6. the purpose of each piece of equipment?
7. agreed upon integration to existing equipment?
8. contact information for support personnel and decision makers?

Have you reviewed our Tips for Successful Deployment?

Do you have all the required equipment?

Do you have the connection blocks, wire and cables required to make connections between different pieces of equipment? Have you thought about how these connections will be accomplished? (For example, surface telephone jacks and telephone line cords to connect RJ11 outputs to cross connect block)

Do you have the necessary mounting hardware, screws, racks, cable ties, Wiremold, etc.

Has all existing site infrastructure and equipment slated for reuse been properly assessed?

Do all parties involved have a common game plan concerning the installation? (for example, wire color codes to use and speaker polarity)

Do you have printed copies of all installation manuals and wiring diagrams?

Do you have all tools necessary to mount, connect and test equipment?

Do you have all tools necessary to access work locations, install equipment, make any necessary adjustments, and perform any necessary tests? Ladders? Scissor Lift?

Have you arranged full unrestricted access to all relevant areas and all relevant equipment?

Will ac power be available where needed? Do you need power strips?

Have you installed all necessary software tools?

Will an Internet connection be available if needed?

Have you reviewed, implemented, and shared all data network requirements?

Will the required people resources, for example the network or telephone system administrator, be available?

Do you have all required network information such as IP addresses, NTP server addresses and a list of job allocated switch ports?

For IP systems, do you have a complete pre-commissioning questionnaire?

Have you reviewed the Best Practices section of this document?

Reusing Existing Infrastructure

It is not uncommon to reuse existing infrastructure in retrofit applications. When doing so, taking the time to properly assess the existing infrastructure, as described later in this guide, will save time and money. Once assessed, the challenge of making reliable connections to the Valcom system need be addressed.

Existing infrastructure often differs from the UTP cabling used with Valcom systems. Existing systems often use heavy gauge, often shielded, speaker wire. Reliably splicing this older type infrastructure to UTP, while leaving connection points available for troubleshooting and maintenance can be challenging. Attempting to terminate heavy gauge wire on traditional punch down blocks can be difficult and unreliable.

Oftentimes, rookie, or resource limited, technicians resort to using Dolphin® Connectors to make a "bean flower". This technique, however, does not facilitate troubleshooting and maintenance and can also contribute to system noise and crosstalk issues.

Better techniques involve using Valcom XP-RFB Retrofit Blocks or VM-150 Screw Terminal blocks.

Bean Flower

OPTION USING VM-150 SCREW TERMINAL BLOCKS

VM-150

HEAVY GAUGE SPEAKER WIRE

UTP INTERCONNECT

VALCOM EQUIPMENT
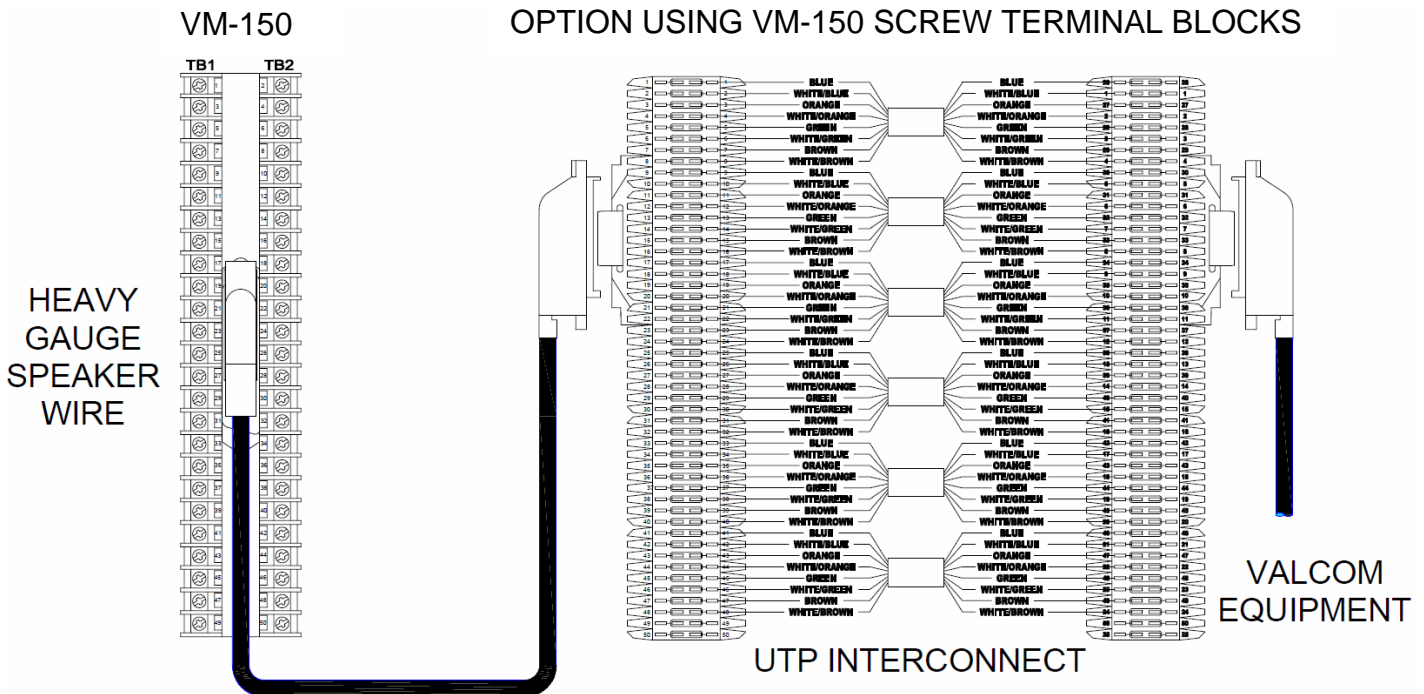
OPTION USING XP-RFB RETROFIT BLOCKS

XP-RFB

XP-RFB

HEAVY
GAUGE
SPEAKER
WIRE

UTP INTERCONNECT

VALCOM
EQUIPMENT
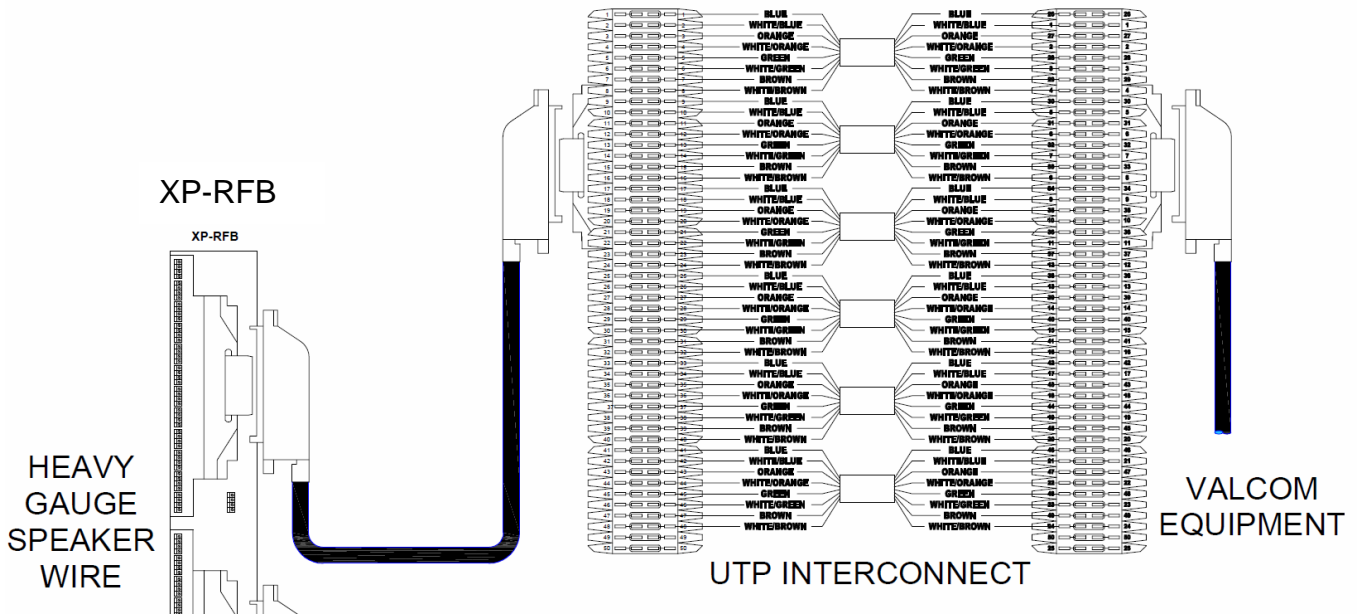
For full size PDFs and AutoCAD drawings, click here.

Shields on existing infrastructure should be:

a) Unterminated
b) Terminated to Ground on the originating equipment end only
c) Used as one of the call switch conductors (if necessary)

Shields used as a conductor for the call switch must be isolated from ground and only used to connect one side of the call switch to its appropriate system termination point.

Reusing Existing Equipment

It is not unusual for new equipment to be interfaced with previously installed, sometimes rather dated, equipment. Contractually defining your scope of work for the newly installed equipment and documenting where your responsibility begins and ends is very important.

For example, when the proper operation of the system that you are providing is dependent on systems that you do not manage, such as telephone systems and facility networks, and those dependencies hinder the project's completion, it is vital that the job contract and/or proposal clearly state that your responsibility is limited to proving that the equipment you've provided is fully functional. You might even contractually agree upon steps to determine this.

There is always a reason why end users contract new equipment upgrades or replacement, it is typically because the old equipment is no longer performing satisfactorily. However, by reutilizing parts of the old system, they potentially introduce troubles to the new system.

1) The accuracy of failing electromechanical clocks with worn gears will not increase because you have replaced the master clock.
2) Intermittent push buttons will still be intermittent when connected to new equipment.
3) Sticking contacts from monitored equipment will still stick when connected to new equipment.
4) Speakers with broken paper cones will not sound better when connected to new amplifiers.
5) Intermittent cabling will still be intermittent when connected to new equipment.

The author recalls a site where relays from Valcom equipment were being utilized to operate 40+ year old electromechanical door locks. These same locks provided a contact closure when the door was unlocked or ajar. These contacts were connected to Valcom equipment to provide indication that the door was unsecured.

Several of the doors would not unlock as designed. However, monitoring the relay outputs of the Valcom equipment verified that the activation contact closures were indeed

operating as designed. Therefore, the locks were at fault. The scope of work was to provide activation for the lock.

On this same site, several doors would intermittently indicate that they were unsecured when they were actually closed and locked. By monitoring the inputs of the Valcom equipment where the "door ajar" contacts were connected, it was obvious that the "door ajar" contacts were sticking closed from time to time, thus causing the fault.

Integrating with non-Valcom Equipment

If you plan to interface your new Valcom solution with non-Valcom equipment, you will need to gather information concerning the desired interface. Depending upon the desired interface, you will need to know one or both of the following.

1) What does the non-Valcom equipment require from the Valcom equipment?
2) What does the non-Valcom equipment provide to the Valcom equipment?

For audio interface, you will need to know:

1) output levels and impedances
2) input levels, sensitivities and impedances
3) physical connection form (XLR connector, RCA jacks, etc)

For contact closure interface, you will need to know the contact voltage/current ratings and operation.

For data interface, you will need details on the data format and any security credentials.

For voltage triggers, you will need to know the voltage type (ac/dc), level, and current capacity.

Tips for successful system deployment

First, read the Best Practices section at the beginning of this document. It is a compilation of time saving observations and site knowledge from thousands of jobsites.

For IP based equipment, also review the information found in the IP6000 Initial Setup Procedure.

Pre-stage and program the head end equipment off-site as much as possible. Programming IP based systems may be accomplished by connecting the various gateways and servers to unmanaged PoE network switches.

Alternately, Valcom IP Gateways and endpoints may be preprogrammed using VIP-102B Sample devices and saving the pre-programmed Snapshot. This method, however, does not allow for pretesting the hardware.

Analog head end equipment will simply need to be connected and powered.

If the physical equipment has been pre-configured as opposed to using sample devices, use a test speaker or a lineman's handset in monitor mode to verify that all analog audio outputs receive audio as expected. Use an Ohmmeter to verify that all relay outputs behave as expected.

Onsite

For retrofits, identify and mark speaker cables with identifying labels.

Pretest all self-amplified speaker lines and old fashioned centrally amplified speaker lines by using a toner or other line level audio source. Individually connect the toner, or audio source to each self-amplified speaker line's tip and ring and to each amplifier input. Note that both self-amplified speakers and old-fashioned amplifiers must be powered for this test. Verify that all speakers are operational.

Using cable ties or tape, group existing speaker cables in increments that match the available outputs and programming of the new Valcom headend speaker outputs. For example, if using 8-port gateways, group speaker circuits in increments of 8 to match the gateways' preprogrammed dial codes. For retrofits, this will be easier if the preprogrammed dial codes sequentially match the physical order of the existing system's cables.

For new and retrofit systems assess speaker wiring following the procedure outlined in the Assessing Field Wiring section of this document.

Install the new head end equipment.

Using a system of cross connect blocks and interconnect wiring, connect the speakers to the new headend equipment. If using 8-port gateways with speakers terminated with 4

pin Molex connectors (speaker and call switch), utilize the 4 pin headers of the 8-port gateways by removing the gateway's screw terminal blocks. These screw terminal blocks may be removed by gently prying them off with a small screwdriver.

Connect all peripheral equipment.

Power the new headend equipment.

For systems with dial tone access (FXS port, office telephone port, telephone tip/ring), use your lineman's handset to place test announcements from each dial tone input.

For systems with station access (FXO port), use an analog station port or other trunk to place test announcements from each dial tone input.

This PA/Intercom Site Commissioning Guidelines document provides tips for verifying system operation.

Documenting the Site

When installing *or maintaining* a system, consider that you, or a new individual may have to come back to the site in many years to provide support.

As previously mentioned, proper and descriptive labeling is the first step to successful system maintenance.

Documenting the details of a system's intended operation is crucial to its long-term support. A detailed overview of the purpose of the Valcom equipment will not only ensure that new staff and support personnel can maintain and make additions to the Valcom system, but also utilize the system to its full capacity. In addition to the detailed overview of the system's intended operation:

1) Document each piece of equipment's location and role. Use MAC addresses, labels and as-builts for positive identification in the future
2) Record each speaker I/O and the areas it serves
3) Record final settings of all adjustable volumes, timers, etc.
4) Record each switch input and its purpose in detail (I/O unit, dial code 303, input 1, activates the "all clear" Play List, which send the all clear event to Facebook, Twitter, groups 909, 200 and 400 and operates I/O unit 304 relay 5 for 2 seconds to unlock the main doors)
5) Record each relay, or other output, and its purpose (I/O unit 304 relay 5 on for 2 seconds unlocks the main doors)

6) Record each audio input and its purpose (Audio Gateway, Dial code 500 input 6 is a source for background music from the CD player labeled "main music")

7) Identify infrastructure destinations by the selected labeling scheme. (Cable A25H routes from Closet 23LR on 66 block 7 to Closet 87RH on 110 block 9 and provides audio and power to zone 67)

8) Record which endpoints are associated with each other (FXS Gateway, dial code 000, channels 1 and 2, are connected to VEADP phones in the security area)

9) Record the intent of each emergency message initiation ICON, dial code or other trigger and the expected response. (what disseminates the message, the timing of the message delivery, where and how the message is delivered and how messages may be cancelled)

10) Describe and photograph termination points as much as possible so that future troubleshooting or adds/moves/changes are easily accomplished

11) Use descriptive labeling in programming tools to support this documentation

12) Record the location of system programming backups (a hosted drive works well in this capacity or ask Valcom to keep archived copies)

Record these details as if preparing for a completely new support person to take over maintenance without any assistance from the original installer.

Keep this documentation up to date as changes are made and provide current copies, as well as programming Snapshots and backups to the owner, Valcom and keep copies for your own future reference.

Redundant Designs

When designing mission critical systems, designers should consider redundancy.

For systems that rely upon a facility network, network redundancy should be implemented. Network redundancy provides a means for network operations to utilize alternate "standby" infrastructure in the event of unplanned network outages.

Such redundancy may be achieved by providing alternate network paths, redundant standby routers, and switches. Should a failure occur in the primary network path, these alternate paths will automatically be available to provide backup routing.

For public address systems, source alternate speakers from different distribution frames served by unique ac power (i.e. different phases, different electrical panels, different circuit breakers). By doing so, if one of the distribution frames loses power every other speaker will still be operational.

IP speakers should be connected such that if a speaker connects to the network switch in distribution frame A, the next IP speaker connects to the network switch in distribution frame B. IP LED sign connections should follow the same pattern.

For analog speakers, wire alternating speakers such that if a speaker connects to an audio source in distribution frame A, the next speaker connects to an audio source in distribution frame B. The 2 distribution frames should have their own power supplies, if necessary.

All system components requiring power should be matched with appropriately sized battery backup (UPS) systems.

Provide multiple ways to invoke public address announcements. SIP, FXS gateways, Interactive consoles and/or microphones should be deployed throughout the facility and should be served from independent distribution frames.

High availability options for system servers such as Application Servers should be incorporated into your design with each server of a high availability pair located in a unique distribution frame.

The More You Know

The Blessing of Battery Backup

Communication systems are a vital part of any enterprise. The importance of such systems grows proportionally along with the size of that enterprise. This is especially true in modern business, healthcare, transportation, government, and educational facilities.

The ability to communicate with students, staff and clients is mission critical.

The importance of communication systems to everyday operations is only eclipsed by their importance in crisis situations. During crisis situations, a daily communication system rapidly gains high status as an Emergency Communication System.

Unfortunately, crisis situations are often accompanied by municipal power failures.

In such situations, communication systems that rely solely on these failed power systems will cease to function and, in their absence, chaos, danger, and loss of life may occur.

Additionally, if failed power is restored in phases, systems that rely upon each other for proper operation may falter and may not automatically recover. For example, devices set for DHCP may not be able obtain an IP address upon power restoration. This will occur if power to the network DHCP server lags power restoration to DHCP dependent network endpoints, or if they require significant boot time upon power restoration. Some DHCP servers may require 10 minutes or more to recover from facility power cycles.

For these reasons, ensuring that mission critical systems such as network components and communication systems are paired with appropriately sized battery backup systems is crucial.

Even if automatic backup generators are available, there will be a brief interruption between municipal power failures and generator startup. This is enough to cause affected systems to reboot thus causing an interruption to service.

How to Make a Public Address Announcement (page)

To make a page, lift a telephone handset and press the desired overhead paging code on the dial pad or press the pre-programmed page button if provided.

WHEN MAKING A VOICE PAGE

1) Hold the telephone mouthpiece near your mouth as if you were in a normal telephone conversation

2) Speak slowly and distinctly in a normal telephone conversation volume level

3) Say your message twice for greater recognition and intelligibility

4) When saying a name or number, i.e., "extension 2930", always enunciate each word individually with a slight pause between numbers, i.e., "extension   two – nine – three – zero". "Zero" is more easily understood than "Oh"

When ending the page, press the "release" button (if provided) or press the hook switch prior to putting the telephone handset onto the cradle to reduce or eliminate handset rattle over the speaker system.

Upgrading Firmware

Please Wait...

Newer is not always better. Firmware changes are implemented for a variety of reasons. Often for reasons that are not relevant to your specific installation.

Even though we test the firmware rigorously, it is best to avoid upgrading firmware unless there is a specific reason to do so.

Blindly upgrading firmware in any system comes with risk, for example, losing power during the process might corrupt the upgrade. Remember the adage "If it ain't broke, don't fix it"

When you DO upgrade firmware, and you have more than one of the same product type to upgrade, do not upgrade all of them at once. Upgrade one, test it, then slowly upgrade the others as you gain confidence in the result.

Before upgrading:

- ✓ Consult with Valcom Support to verify firmware compatibility and proper upgrade steps. Occasionally, firmware updates must be applied in a specific order.

- ✓ Always perform *and download* a system backup.

- ✓ If possible, verify proper device operation.

- ✓ Be aware that the Valcom system may be out of commission during the upgrade process.

In the case of Valcom servers:

- ✓ Perform the steps above.

- ✓ ***Before you begin***, unplug the server power supply. Be certain that the server continues to operate for 10 to 15 minutes. After testing this, plug the power supply back in for at least 10 minutes before upgrading.

- ✓ ***After you apply the firmware***, be patient. The server will be unresponsive during the upgrade. This is normal. Wait a minimum of 10 to 15 minutes after applying the upgrade before attempting to access the server. DO NOT INTERRUPT POWER.

Wiring Facts and Myths

1) In most cases, transporting line level signal from a source to a significantly higher impedance load is easily accomplished with a small gauge pair of wires. For efficient signal transfer, the source output impedance should <= the load input impedance.

2) The role of wire gauge becomes a factor when directly powering speakers or equipment and is a function of both the current required by the powered device(s) and the cable distance between the power source and the load(s). Most manufacturers publish pre-calculated guidelines for their equipment. These guidelines are typically safe to use and most include assumptions that err to the side of caution. However, no single chart can describe all of the possible circuits that might result from powering multiple devices in parallel. There can be a big difference between having the device load evenly spaced along a length of wire and having that same device load at the end of the length of wire.

3) When using spare pairs in multi-pair cable to power devices at various distances from the source, there are a couple of schools of thought. Some say that twisting the unused pairs together to emulate a higher gauge pair for the entire cable length is best. Some say that using each spare pair to source an equal number of adjacent devices is best. In practice, both techniques yield an equivalent result with the latter resulting in less connectivity issues. Whatever you do, when combining pairs, use one conductor of each combined pair for the positive side of the source signal/voltage and one conductor for the negative side. Never split pairs or use whole pairs as a single conductor.

Combining Pairs to Increase Effective Wire Size

If you twist two wires together, each would carry half the current, so you effectively increase the gauge. American Wire Gauges go down by about 10 for every factor of ten in cross-sectional area. If you had ten 20 AWG wires connected in parallel, they could effectively carry as much current as one 10 AWG wire. With two 20 AWG wires, you'd have the equivalent of one 17 AWG wire. ***Smaller AWG = larger conductor size = more cross-sectional area = less resistance for any given length.***

Some "rules of thumb":

40 AWG copper wire has about an Ohm of resistance for each foot. By the rule above, 30 AWG copper wire would have an Ohm of resistance for every ten feet, and 20 AWG an Ohm of resistance for every 100 feet.

Combining 2 conductors of the same AWG wire will result in a drop of 3 AWG size increments (two 24 AWG conductors combined = one 21 AWG conductor)

Combining 3 conductors of the same AWG wire will result in a drop of 5 AWG size increments (three 24 AWG conductors combined = one 19 AWG conductor)

Note that connecting wires in parallel may work at dc or low frequency ac, for audio, RF, or other purposes, it's best to use a single pair.

Twisting Pairs Together to Increase Wire Gauge

White/Blue ——————→ Audio TIP of Speaker or Horn
White/Orange ————→
White/Green —————→ Ground (+) of Speaker or Horn
White/Brown —————→

Blue/White ——————→ Audio RING of Speaker or Horn
Orange/White ————→
Green/White —————→ -24Vdc (-) of Speaker or Horn
Brown/White —————→
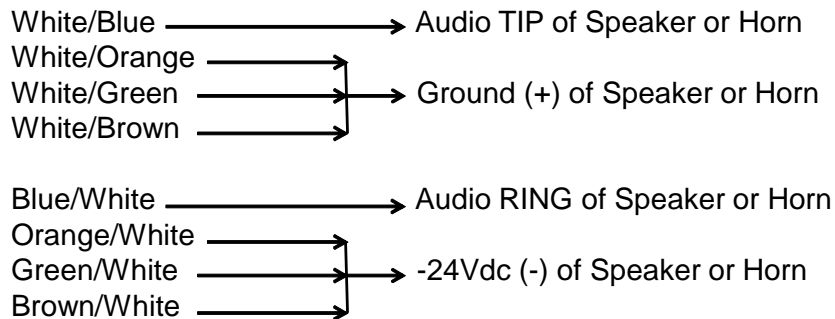
Valcom suggests using Cat 3 or higher UTP for self-amplified and 45-ohm speaker connections.

Solid vs. Stranded

Some manufacturers will state that stranded wire has better high frequency response, and they are correct. However, this is true only at frequencies way above the audio thresholds of the human ear. The benefits of stranded wire are in its flexibility and its ability to better transport very high frequencies in the megahertz range (see skin effect). Therefore stranded/braided wire is used to connect lightning rods to Earth ground – lightning strikes result in a very high frequency current spike. The disadvantage of stranded wire is the inability to make stable connections to various types of connectors and cross-connects.

Relationship Between Source Capacity and Wire Size

You cannot compensate for inadequately sized cabling by choosing a higher capacity source. Having an adequately sized source (amplifier, power supply) and having adequately sized cabling are equally important, yet unrelated, design criteria.

Audio Fun Facts

There are many misconceptions about the amount of power required from speakers and horns. This is largely because speakers used with power amplifiers often have tap settings of 5 or higher watts. Adjusting the power amplifier output level to account for wiring losses and adequate speaker volume results in much less speaker/horn power consumption.

Here are some fun facts:

The goal of a public address system, for voice announcements, is to be *≈6db* louder than an area's ambient noise level at average listener's ear level.

The goal of a public address system, for tones, is to be *≈3db* louder than an area's ambient noise level at average listener's ear level.

Most speakers and horns use very little power:

After source adjustment for signal loss in wiring and adequate sound level, a speaker in a typical indoor office/classroom environment uses ≈8 mW (thousandths of a Watt) of power despite being tapped for much higher power. Refer to Amplifier Facts and Myths.

A +3 dbspl change is barely perceptible

A +10 dbspl change is considered twice as loud

A +20 dbspl change is considered four times as loud

Every time you double power to a speaker or horn you gain ≈ 3db

Every time you halve power to a speaker or horn you lose ≈ 3db

Unless an amplifier is overtaxed or failing, replacing it with an amplifier to a higher power model will not change the volume of the existing system speakers or horns.

Adjusting speaker taps on an old fashioned centrally amplified speaker will effect the sound level on all other speakers and horns on the same cable.

By the Inverse Square Law:

Sound level is diminished by ≈6db as distance from a sound source doubles

Sound level increases by ≈6db as distance from a sound source halves

Decibels and Sound Pressure

The Decibel (dB) does not refer to any specific level but is a logarithmic ratio between 2 values of electrical signal, power or sound pressure.

Since the dB expresses relative level change, it is used to describe both acoustic volume levels and ratios between signal voltages, signal currents and signal power levels.

With respect to signal power ratios:

dB loss or gain = 10 x Log (Input Power/Output Power)

With respect to signal voltage or current ratios:

dB loss or gain = 20 x Log (Input Level/Output Level)

Acoustic dB, or dBspl, refers to an acoustic pressure level (a.k.a. volume or loudness) level relative to a 0-dBspl reference. 0-dBspl refers to 20 μPa of pressure.

Normal human hearing can perceive 0-dBspl and finds 120-dBspl to be painfully loud.

The dB changes in a system are all interrelated. In other words, discounting system losses and limitations, a 6-dB increase in signal voltage will result in a 6-dB increase in power which, when applied to speakers or horns, will result in a 6-dBspl increase in sound pressure (volume).

For example, consider an 8 Ohm speaker being sourced from an amplifier. For discussion purposes, let's assume that the speaker will produce 96-dBspl when 1 watt of power is applied. The signal voltage to produce 1 watt of power into an 8 Ohm load is:

$$E = \sqrt{PR}$$

$$E = \sqrt{1 * 8}$$

$$E = \sqrt{8}$$

$$E = 2.828 \, volts$$

Now we'll double the signal voltage to 5.656 volts and calculate our expected power level:

$$P = \frac{E^2}{R}$$

$$P = \frac{5.656^2}{8}$$

$$P = 4 \, watts$$

Does the dB change of signal voltage correlate to the dB change of power?

We started with 1 watt of power, which corresponded to 2.828 volts of signal.

We doubled the signal voltage to 5.656 volts, which resulted in 4 watts of power.

Yes, they are both 6-dB changes:

$$20 \, Log(\frac{5.656 \, volts}{2.828 \, volts}) = \text{6-dB}$$

$$10 \, Log(\frac{4 \, watts}{1 \, watt}) = \text{6-dB}$$

And likewise, the dBspl output from the speaker would increase by 6-dBspl.

So, you see, by using dBs, you can simply add and subtract gains and losses in a system without regard for unit of measure conversion. A 3-dB input into an amplifier, which has a gain of 20-dB will result in a 23-dB output.

It's important to note that if the impedance values used to calculate dB loss are different for the 2 values forming the ratio then dB values calculated from voltage or current ratios must factor in the square root of the varying impedances: Click here for examples.

$$When \; impedances \; are \; equal: dB = 10 \, Log \frac{P2}{P1} = 20 \, Log \frac{E2}{E1} = 20 \, Log \; \frac{I2}{I1}$$

$$When \; impedances \; are \; NOT \; equal: dB = 10 \, Log \frac{P2}{P1} = 20 \, Log \frac{E2\sqrt{Z1}}{E1\sqrt{Z2}} = 20 \, Log \; \frac{I2\sqrt{Z2}}{I1\sqrt{Z1}}$$

In our example, the speaker was rated by the manufacturer to produce 96-dBspl. However, that's not really enough information to be useful.

dBspl output from a speaker decreases as you move away from the speaker and increases as you move towards the speaker. Therefore, we cannot simply state that a speaker produces 96-dBspl, we also have to specify the distance directly in front of the speaker at which 96-dBspl is produced. And in reality, because speakers don't produce the same sound pressure for every frequency at a given power input, we also have to specify the frequency used for the measurement.

Published dBspl outputs are typically measured at 4 feet or 1 meter and 1 kHz.

So how do we determine the sound pressure at other frequencies? You'd have to measure the response of the speaker at each frequency while maintaining the specified power input. Fortunately, you need not be concerned; paging speakers are designed to provide adequate dBspl at all of the frequencies required for voice and background music reproduction.

And another question would be, how do we determine the dBspl at various distances in front of the speaker? We use the [Inverse Square Law](#).

$$dBspl \ loss \ or \ gain = 20 \ Log \left(\frac{Distance \ of \ Interest}{Reference \ Distance}\right)$$

As the sound pressure travels away from the speaker, it spreads out over an area that increases in proportion to the square of the original measurement distance (typically 1 meter). In practice you'll find that doubling the distance results in ≈ 6-dBspl of loss.
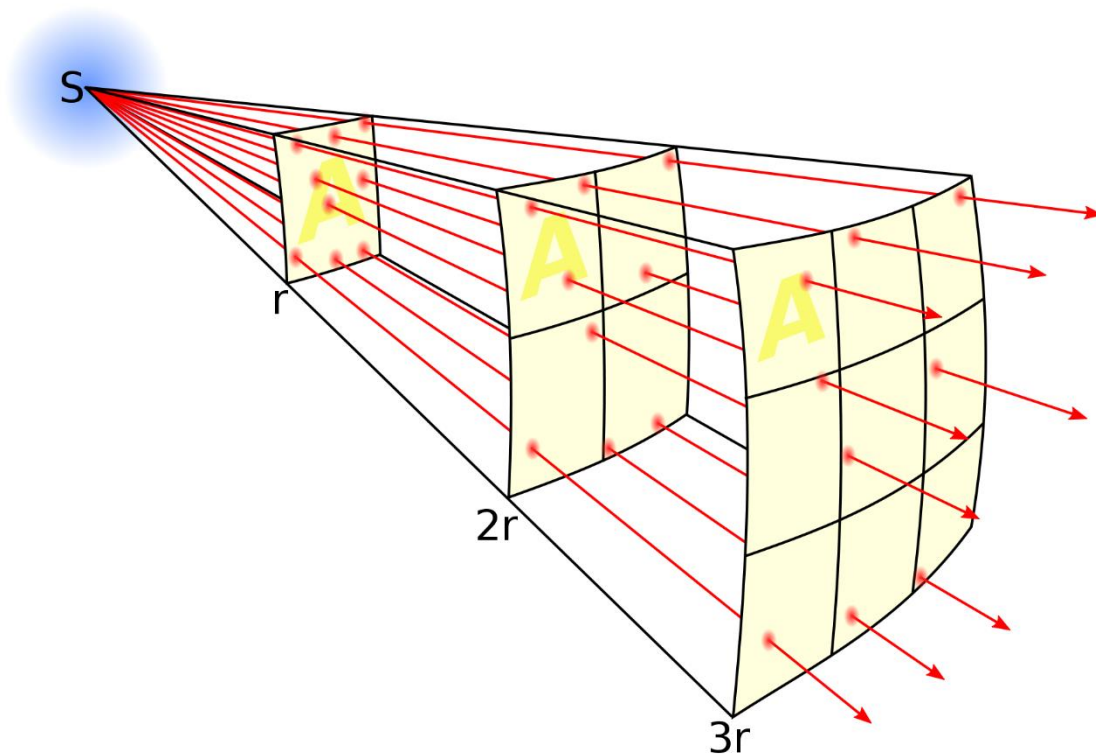
It's important to note that the dBspl also diminishes as listeners move away from the speaker's direct sound. The angle away from center at which the sound diminishes by 6-dBspl is known as the dispersion angle. Where the dispersion angles of 2 in phase speakers meet, the sound pressure from one will boost the sound pressure from the other and will result in a 3-dBspl boost. You can visualize this by picturing the streams of 2 garden hoses merging. The water pressure where the streams meet is boosted.

This is why symmetrical, properly spaced speakers and consistent phasing (using the same polarity on all speaker connections) are crucial for even sound in an area.

In the sound reinforcement/public address system industry, to effectively communicate changes in audio power level, it's important to become comfortable with decibels (dB).

Using decibels to express ratios of quantities provides for using simple addition and subtraction of various gain or loss stages in a system to determine total system loss or gain. This is especially true when this loss or gain spans many orders of magnitude.

Ratio of Output/Input	dB value if the ratio is a Voltage or Current ratio	dB value if ratio is a Power ratio
.0000001	-140	-70
.000001	-120	-60
.00001	-100	-50
.0001	-80	-40
.001	-60	-30
.01	-40	-20
.1	-20	-10
1	0	0
10	20	10
100	40	20
1000	60	30
10000	80	40
100000	100	50
1000000	120	60
10000000	140	70

Decibels are also used to express the amplitude of various signal parameters such as signal voltage or current relative to a given reference level. Once again, for power ratios:

dB change is simply calculated as:

$$dB = 10Log\frac{P2}{P1}$$

For example, if a gain stage or whole circuit doubles the power of a signal, then:

$$dB = 10Log\frac{2}{1} = 10Log\ 2 = 3\ dB$$

So, any time power is doubled, it's a 3-dB change. What about halving the power?

$$dB = 10Log\frac{1}{2} = 10Log\ 0.5 = -3\ dB$$

Why would you want to know this?

Because if you are reading this, you are probably working with sound producing equipment, or, you are really bored (or both).

Anyway -

As previously stated, human hearing in not linear and *twice the power is not perceived by human hearing*[1] *as [twice as loud](#).*

As rough rules of thumb:

A 3-dB$_{spl}$[2] gain is perceptible to most people.

A 6-dB$_{spl}$ gain is perceived is easily discernable to most people.

A 10-dB$_{spl}$ is *perceived* to be approximately twice as loud.

A 20-dB$_{spl}$ gain is perceived to be about *4 times as loud*.

A 40-dB$_{spl}$ gain is perceived to be about *16 times as loud*.

Now, knowledge of dBs expressing a power ratio can be quite useful. If you have an amplifier connected to a speaker and the amplifier is providing 1-Watt of audio power to the speaker, assuming the speaker is mechanically capable, doubling the audio power will result in a 3-dB$_{spl}$ increase in sound pressure from the speaker, *quadrupling* the audio power will result in a 6-dB$_{spl}$ increase in sound pressure from the speaker. *This is good to know!*

It's also good to know that these increases are with respect to a specified distance from the speaker. Stating that a speaker produces 96-dB$_{spl}$ with 1-Watt of power applied is meaningless without specifying an "on axis" distance at which the sound pressure will occur (i.e. 96-dB$_{spl}$ at 4 feet).

Tie this knowledge together with the fact that in typical environments, paging audio that is 6-dB$_{spl}$ above an area's ambient noise is easily discerned and you know that you will need your speakers to produce enough sound pressure to the listeners to overcome the area's ambient noise by 6-dB.

But it's not good enough to produce adequate sound pressure near the speaker, you have to keep in mind that sound diminishes as a function of the square of the distance from its source (inverse square law) and you'll have to account for the sound pressure lost as a function of the listeners' distance from the speaker. Once again, here is the Inverse Square Law:

$$dBspl = dBspl\ at\ Reference\ distance - 20Log\frac{Distance\ of\ interest}{Reference\ distance}$$

Generally, the reference distance dBspl is measured at 4 feet or 1 meter from the speaker.

Example:

If you have a listener that will be:

   a) 10 feet in front of a speaker that produces 96-dB$_{spl}$ at 4 feet
   b) In a room with an ambient noise level measured to be 75-dB$_{spl}$

The speaker will have to produce 6-dB$_{spl}$ above 75-dB$_{spl}$ at a distance of 10 feet for adequate sound level.

How much power must we provide to the speaker?

The listener will need to receive 81-dB$_{spl}$ (75-dB$_{spl}$ + 6-dB$_{spl}$ = 81-dB$_{spl}$ of sound pressure)

We can calculate how much sound pressure will be lost between the speaker location and the listener from the Inverse Square Law formula:

Loss at 10 feet = $20 Log \frac{10\ feet}{4\ feet} = 7.96\ dBspl$

So, the speaker needs to produce 7.96-dB$_{spl}$ plus 6-dB$_{spl}$ above the 75-dB$_{spl}$ ambient noise level to deliver 81-dB$_{spl}$ to the listener. That's 88.96-dB$_{spl}$ at 4 feet.

Since the speaker is rated to produce 96-dB$_{spl}$ at 4 feet with 1-Watt of power applied, and we only need 88.96-dB$_{spl}$ at 4 feet for this scenario, we can obviously drive the speaker at *less* than 1-Watt.

How many dB less that the rated output is needed? 96-dB$_{spl}$ - 88.96-dB$_{spl}$ ≈ 7-dB$_{spl}$

There are certainly way to mathematically figure out exactly how much power it will take to cause the speaker to provide 7-dB$_{spl}$ below its rated output, but ***now the power of understanding the relationship between dBs and power ratios to provide quick math-free estimations becomes evident:***

We know that halving or doubling power results in a 3-dB change - $(10\ Log\ \frac{1}{2} = -3dB)$

Since 1 Watt provides 96-dB$_{spl}$ at 4 feet

Halving the power to ½ Watt will provide ≈ 93-dB$_{spl}$ at 4 feet

Halving the power again to ¼ Watt will provide ≈ 90-dB$_{spl}$ at 4 feet

90-dB$_{spl}$ is close enough to 88.96-dB$_{spl}$ so we will provide ¼ Watt of power to our speaker!

[1]Perceived loudness perception is also a function of frequency but that's a different discussion

[2]We *use the term dB$_{spl}$ to designate the dB value as related to sound pressure*

Example of finding the dB loss in a circuit by summing the dB losses in each stage

Using dBs to express the gains and losses through various stages of a circuit or system facilitates calculating total gain or loss as they simply sum. Here is an example with a simple series parallel resistor circuit using simple values. The gain or loss is independent of the signal or voltage level applied, so is this example will arbitrarily use a 100-volt source; however, any source would result in the same dB gains or loss.
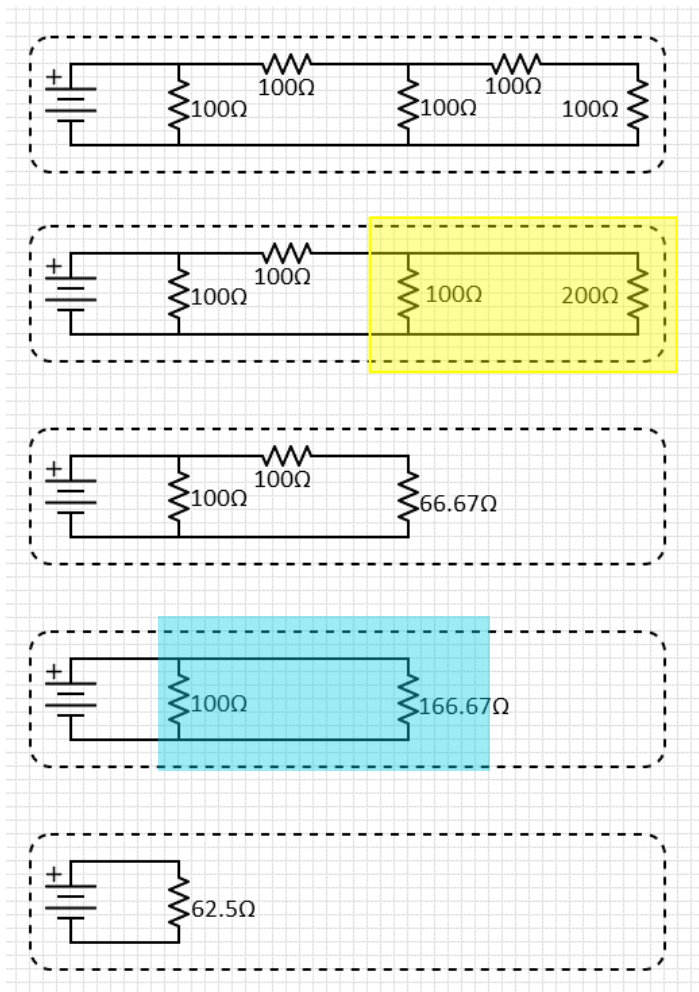
1) Combine the rightmost 100Ω resistors.

2) 100Ω + 100Ω = 200Ω

3) Find the parallel equivalent of the right most parallel resistors

4) $\dfrac{1}{\frac{1}{100}+\frac{1}{200}} = 66.67\Omega$

5) Combine the rightmost 100Ω and 66.67Ω resistors.

6) 100Ω + 66.67Ω = 166.67Ω

7) Find the parallel equivalent of the right most parallel resistors

8) $\dfrac{1}{\frac{1}{100}+\frac{1}{166.67}} = 62.5\Omega$
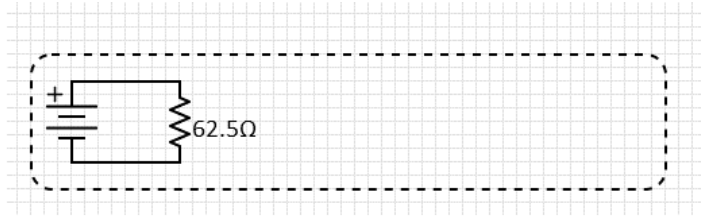
This single 62.5Ω resistor is equivelant to the 5-resistor series/parallel circuit we started with. If the power source is 100 Volts, then the current being sourced from the supply is:

$$I = \frac{E}{R} = \frac{100V}{62.5\Omega} = 1.6\ Amps$$

And the power dissipated across the 62.5Ω resistor is:

$$P = \frac{E^2}{R} = \frac{100^2}{62.5\Omega} = 160\ Watts$$
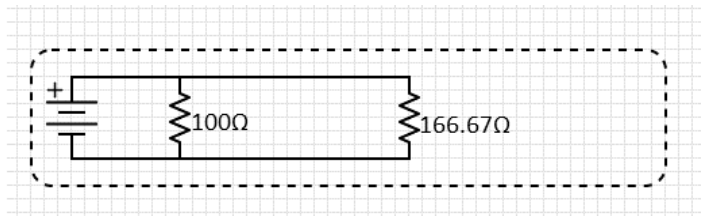
Expanding the circuit out from the left, we can easily find the current through the 166.67Ω resistor simply because we know that the voltage across the resistor is 100V:

$$I = \frac{E}{R} = \frac{100V}{166.67\Omega} = 0.6\ Amps$$

And the power dissipated across the 166.67Ω resistor is:

$$P = \frac{E^2}{R} = \frac{100^2}{166.67\Omega} = 60\ Watts$$

The dB loss in this stage of the circuit is:

$$10\ Log\ \frac{60W}{160W} = -4.26\ dB$$
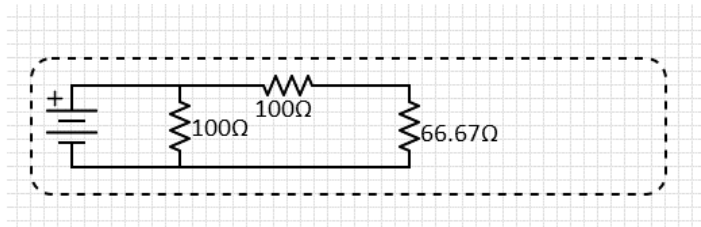
Continuing to expand the circuit out from the left, we can easily find the voltage across the 66.67Ω resistor simply because we know that the current passing through the resistor is 0.6 Amps:

$$E = I \; x \; R = 0.6A \; x \; 66.67\Omega = 40 \; volts$$

And the power dissipated across the 66.67Ω resistor is:

$$P = \frac{E^2}{R} = \frac{40^2}{66.67\Omega} = 24 \; Watts$$

The dB loss in this stage of the circuit is:

$$10 \; Log \frac{24W}{60W} = -3.98 \; dB$$
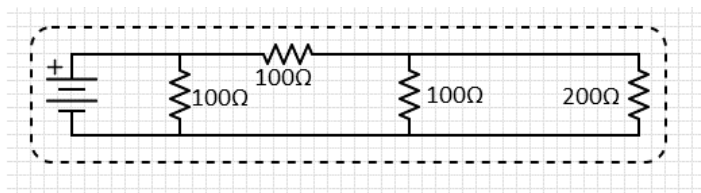
Continuing to expand the circuit out from the left, we can easily find the current through the 200Ω resistor simply because we know that the voltage across the resistor is 40V:

$$I = \frac{E}{R} = \frac{40V}{200\Omega} = 0.2 \; Amps$$

And the power dissipated across the 166.67Ω resistor is:

$$P = \frac{E^2}{R} = \frac{40^2}{200\Omega} = 8 \; Watts$$

The dB loss in this stage of the circuit is:

$$10 \; Log \frac{8W}{24W} = -4.77 \; dB$$
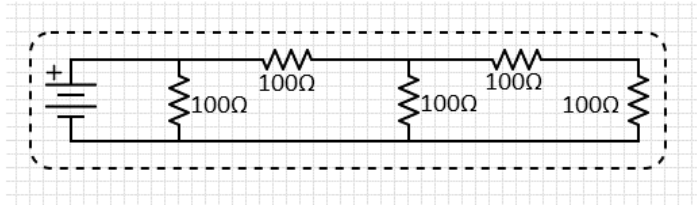
Now, back to our original circuit, we can easily find the voltage across the right most 100Ω resistors simply because we know that the current passing through the resistors is 0.2 Amps:

$$E = I \times R = 0.2A \times 100Ω = 20 \ volts$$

And the power dissipated across each of the rightmost 100Ω resistors is:

$$P = \frac{E^2}{R} = \frac{20^2}{100Ω} = 4 \ Watts$$

The dB loss in this stage of the circuit is:

$$10 \ Log \frac{4W}{8W} = -3.01 \ dB$$

The dB loss from beginning to end of the circuit is:

$$10 \ Log \frac{4W}{160W} = -16.02 \ dB$$

The sum of the dB losses in each stage, (-4.26-dB + -3.98-dB + -4.77-dB + -3.01-dB) of the circuit equals the sum of the total circuit loss (-16.02-dB).

It's important to note that if the impedance values used to calculate dB loss are different for the 2 values forming the ratio then dB values calculated from voltage or current ratios must factor in the square root of the varying impedances: Click here for examples.

By using the **power** dB formula, we concluded that the total dB loss across the circuit is 16.02 db. We should be able to calculate the same loss by comparing the circuit's starting voltage and impedance (100V across a 62.5Ω load impedance) to the final point of loss (20 volts across 100Ω)

$$20 \ Log \frac{20\sqrt{62.6}}{100\sqrt{100}} = -16.02 \ dB$$

Ceiling Speaker Spacing

For ceiling speakers with a 90° dispersion angle, spacing speakers such that the dispersion angles meet at average ear height results in very even coverage. This is because the overlapping coverage provides a 3 dBspl boost to in phase sound. This method is often utilized in systems that will feature background music (so listeners don't hear the music fade out between speakers) or where audio coverage is critical. For example, you'll typically observe denser speaker coverage at noisy airport gate areas.

The -6 dBspl levels from each speaker sum when they merge. The result is a -3 dBspl level relative to the direct sound.
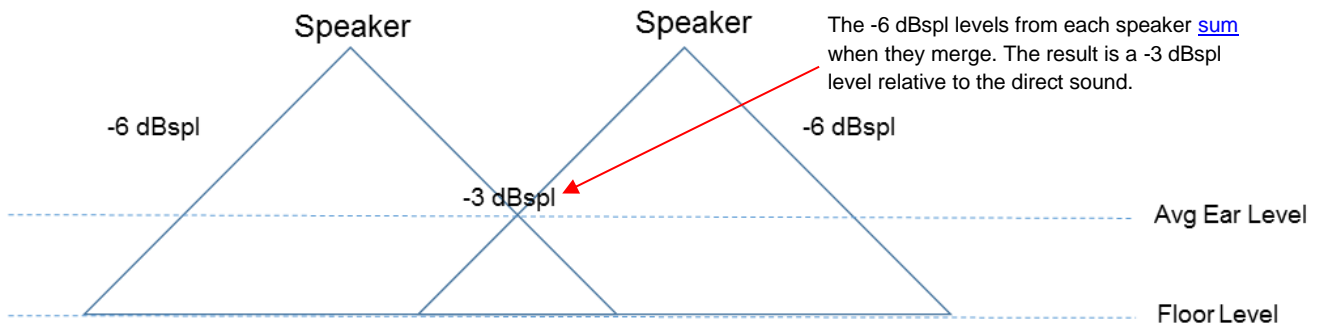
However, for economic reasons, spacing ceiling speakers at twice the mounting height is the widely accepted convention. This results in the dispersion angles meeting at floor level.

Spacing at 2x mounting height
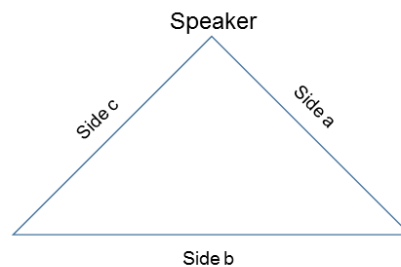
Because the speaker is mounted in parallel with the floor and angle ∠ac is 90° (our dispersion angle), sides a and c have equal lengths. This results in a 90°/45°/45° triangle. If we draw a straight line that splits the 90° angle in half, we end up with 2 smaller 90°/45°/45° triangles and side b is split in half, we'll refer to it as side e.
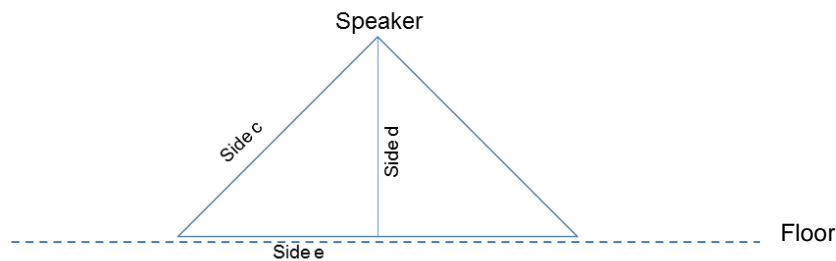
In reference to the previous diagram, if the speaker is mounted at 8 feet, then side d = side e = 8 feet and the speaker spacing for the dispersion angles to meet at floor level will be twice side e and therefore twice the mounting height, or 16 feet in this case.

<u>Spacing so that dispersion angles meet at ear height</u>

If we want the dispersion angles to meet at average ear height (4 feet*), then we simply pretend that the floor is 4 feet higher.
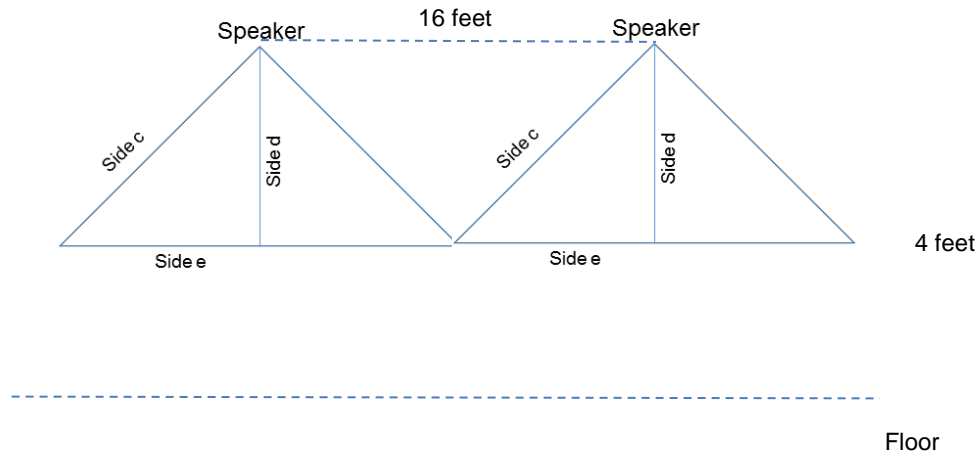
If the speaker is mounted at 12 feet, we calculate as if it were mounted 4 feet lower (8 feet) then side d = side e = 8 feet and spacing becomes twice side e or 16 feet.

With this method, the dispersion angles of adjacent speakers overlap at 4 feet* above the floor. 4 feet is the commonly used as the average listener's ear height*. The sound from each speaker is 6-dBspl lower (definition of dispersion angle) at the overlap point. The in-phase sound pressures sum and the net result is a 3-dBspl boost.

In this way, the sound will only vary by 3-dBspl anywhere in the room.

* For typically standing audiences use 5 feet, for typically seated audiences use 3.5 feet, for average use 4 feet.

Estimating the number of ceiling speakers required for an area:

Critical areas such as airport gates require particularly good audio coverage. For these areas divide the Total Area Square Footage by [(Mounting Height – Average Ear Height) x 2]^2. Speaker spacing = [(Mounting height – Average Ear Height) x 2].

For economical coverage divide the Total Area Square Footage by [Mounting Height x 2]^2. Speaker spacing = [Mounting Height x 2].

If using metric units of measure square footage becomes square meters.

Amplifier/Power Facts and Myths

Loudness is subjective, that is, different people perceive loudness in different ways. In science and engineering, the decibel (dB) is used to describe the ratio of 2 values of power or intensity. Therefore, when we discuss the difference between 2 levels of sound intensity, we speak in terms of dB change.

Decibel changes are logarithmic as opposed to linear. The logarithmic scale expresses wide ranging quantities in simple terms. Being logarithmic, a doubling of volume is not equal to a doubling of dB.

The following chart shows how dBs relate to level and commonly perceived loudness changes.

Change in dB	Change in Signal Voltage	Change in Watts	Change in Loudness
+3	1.4 x	2 x	1.2 x
+6	2 x	4 x	1.5 x
+10	3.16 x	10 x	2 x
+20	10 x	100 x	4 x
+40	100 x	10,000 x	16 x
+80	10,000 x	100,000,000 x	256 x

Notice that a doubling of loudness requires an increase of 10dB, which translates into <u>ten times</u> more power from the amplifier.

There are many myths about how much power is required for speakers and horns.

How much power is actually required?

Speakers and horns are rated in terms of how much sound pressure (dBspl) they will produce at a reference frequency[1] (typically 1 KHz) and a reference power level (typically 1 Watt).

Using this information, it's easy to calculate how much sound pressure (dBspl) they will produce at the reference frequency when other power levels are applied.

Within the mechanical constraints of the speaker or horn, sound pressure output (dBspl) for any single frequency relates to the electrical power applied per the following:

dBspl loss or gain = 10 x Log (Power of Interest/Reference Power)[2]

As an example:

A typical speaker specification states that when 1 Watt of 1 KHz signal is applied, an 8-inch (20.32 cm) round speaker will produce approximately 97.5-dBspl of on axis sound at 1 meter.

Using the power gain/loss formula, you'll notice that halving or doubling power results in a 3dB change:

1 Watt	97.5-dBspl@ 1 meter
½ Watt	94.5-dBspl@ 1 meter
¼ Watt	91.5-dBspl@ 1 meter
1/8th Watt	88.5-dBspl@ 1 meter
1/16th Watt	85.5-dBspl@ 1 meter
1/32nd Watt	82.5-dBspl@ 1 meter
1/64th Watt	79.5-dBspl@ 1 meter
1/128th Watt (7.8 mW)	76.5-dBspl@ 1 meter

Considering that voice paging should be at least 6-dBspl above the ambient noise level, even with 7.8 milliwatts (mW) of power, the speaker output level is adequate for a typical quiet office.

The reason that old fashioned central 100/70.7/25-volt amplifiers produce so much power is because much of it is lost in the cabling and produces heat instead of sound. It's common to see 5-Watt 8-inch (20.32 cm) speakers specified for office areas. In reality, once the amplifier volume is set to compensate for cable losses and to provide adequate speaker volume, the amount of power actually applied to the speaker is in the milliwatts (mW) range.

Speakers and horns are rarely, if ever, driven to rated capacity.

Valcom's approach is to build an amplifier into each one-way speaker or horn to eliminate the loss between the amplifier and the speaker cone. This approach provides for the availability of full audio output and also allows point of amplification volume control[3]. Integrated amplifiers are powered by dispersed dc power supplies thus eliminating concerns for long cable loss and making self-amplified a better choice for large systems.

As an added benefit, Valcom one-way self-amplified speakers connect with inexpensive UTP cabling. The combination of UTP cabling, line level differential audio signals and differential inputs, cancels most interference from other low voltage cables and nulls potential crosstalk into surrounding low voltage cables.

These benefits are never realized in an old fashioned 100/70.7/25 volt centrally amplified system.

In addition, a superior design requires minimizing single points of failure. An amplifier per speaker or horn results in the ultimate redundancy!

Who cares about the dBspl at the reference distance?

In most cases, system designers are not often interested in the amount of sound pressure (dBspl) at some reference distance; they are interested in how much sound pressure will reach the listeners[3].

The formula for determining dBspl loss as you move away from the known reference distance is ((20 x Log (Distance of Interest / Reference Distance)). This results in approximately 6-dBspl of loss for every doubling of distance. This is "on axis" loss. There are many other factors, such as dispersion angle, that must be considered.

As listeners move out of the "on axis" sound produced by a speaker or horn, the dBspl from that speaker or horn will diminish. The angle at which it diminishes by 6-dBspl is known as the dispersion angle.

When in phase sound pressure from two or more speakers or horns meet, the sound pressures boost each other – if they meet at the dispersion angle, the resulting sound pressure level will enjoy a 3-dBspl boost and will only be 3-dBspl below on axis sound levels – barely noticeable.

[1] Note that speakers and horns never have a "flat" frequency response. The dBspl output vs power input will vary at different frequencies.

[2] This power gain or loss formula ((dBspl loss or gain = 10 x Log (Power of Interest/Reference Power)) also results in another interesting fact, doubling audio signal power only results in a 3-dB gain. For example, a speaker sourced with 10 Watts of audio signal will produce 3-dBspl more sound pressure than if it were sourced with 5 Watts of audio signal. Likewise, halving the audio signal power will only result in a 3-dBspl decrease. Human hearing rarely detects a 3-dBspl change, usually detects a 6-dBspl change and perceives a 10-dBspl change as twice as loud. For example, 10 Watts of audio is perceived as twice as loud as 1 Watt of the same audio.

[3] Speakers with integrated volume controls and amplifiers significantly simplify system audio balancing. By designing with an adequate number of speakers or horns, final adjustment of the sound level is as simple as turning the volume controls.

Expanding Existing Analog Speaker Systems

Public address systems often grow along with the facilities they serve. Expanding existing systems is easily accomplished if you understand what you are working with.

Talkback systems should not be expanded to more than 2 talkback speakers per output, and because each speaker acts as a microphone, perform best in relatively quiet environments. Having multiple speakers picking up background noise and speech can make it difficult to discern the audio from the person with whom you are conversing.

Expanding old fashioned centrally amplified systems that use speakers with step down transformers (i.e. 25/70.7/100 volt) simply requires that you know:

- the voltage and capacity of the amplifier channel you will be adding to
  - look for labeling on the amplifier or seek a manual for the amplifier
- the total wattage of the existing speakers already connected to that channel
  - you will need to measure the speaker lines with an impedance meter and calculate the wattage load based upon the impedance measured and the speaker/amplifier voltage rating (unless the impedance meter does the calculation automatically)
- the total wattage of the new speakers you will be adding

Most commercial audio equipment is not really intended to operate at fully rated capacity for long periods of time. So, it is a great idea to leave 20% headroom.

For example, if you have a 100-watt 70.7-volt amplifier channel, you should only really connect 80 watts of speaker load.

So, suppose you have a 100-watt 70.7-volt amplifier channel and it is currently connected to a 60 watt speaker load, you can add 20 watts of additional speaker load (the sum of the tap settings of your new speakers could be up to 20 watts.

What it you have no more wattage capacity available?

- Buy another amplifier
- Redistribute your speaker load
- Add a Valcom V-1095 Pre-amp Expander and as many Valcom One Way Self Amplified speakers, and their associated power supplies, as required.

Whether adding on to an old fashioned centrally amplified system or not, **Valcom One Way Self-Amplified** systems are extremely easy to expand by simply adding additional power supplies where required. If the number of your connected speakers are going to exceed 150 (less in some cases), simple inexpensive preamp repeaters, like the Valcom V-1094A Pre-amp Expander, may be used to refresh the audio and allow the addition of 150 more speakers*. Simple*.

Transformers- the good, the bad and the ugly

Transformers play a major role in analog signal distribution systems. Simple transformers have a single primary and a single secondary winding. More complex transformers will have multiple primary and multiple secondary windings.

Autotransformers have a single multi-tapped winding.

The windings loop around the transformer core. Each complete loop is called a turn. The signal voltage that occurs across the secondary winding is proportional to the number of primary winding turns to secondary winding turns. (i.e. 100 primary turns and 50 secondary turns would be a ratio of 50:100 or 1:2 and the secondary would measure half of the primary's signal.

Click to visit Wikipedia
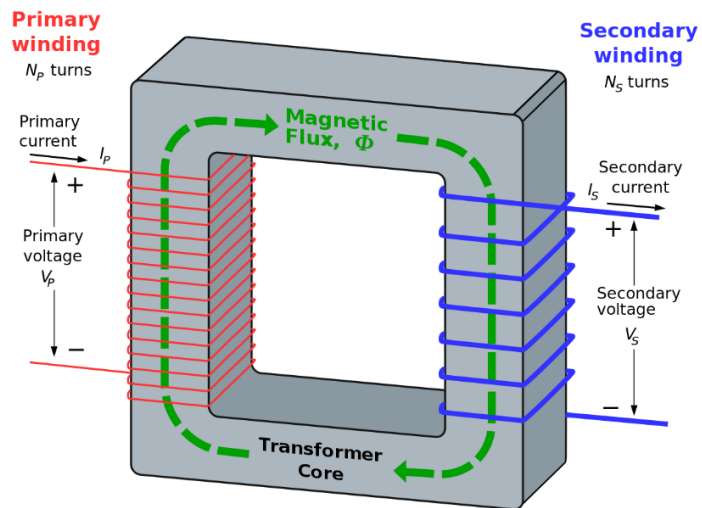
This is how transformers are used to raise and lower signal voltage. The current in the primary vs secondary winding, however, is inversely proportional to the "turns ratio". In this way, discounting transformer losses, the power delivered into the primary winding is the same as the power available from the secondary winding. These formulas are widely available on the Internet.

Transformers with separate primary and secondary windings deliver power and also provide electrical isolation between systems to alleviate reference point disparities or "ground loops".

Some transformers, referred to as 1:1 or isolation transformers, are used solely for this purpose and have the same number of primary and secondary turns. They do not alter the signal level, however, provide electrical isolation.

The alternating primary signal induces magnetic flux in the transformer core. Faraday's law tells us that a changing magnetic flux will induce an electromagnetic field in a coil, in this case the secondary coil.

The bad and the ugly. . .

The electromagnetic field of a transformer may couple noise (undesired signal) into surrounding conductors. An example would be 60 Hertz noise induced in audio pairs routed near florescent light ballasts, which are power transformers, or audio equipment picking up induced noise from close ac power transformers.

The importance of multicast in Valcom VoIP systems

Valcom VoIP systems require properly configured multicast:

1) As the primary method of discovering new endpoints via the VIP-102B IP Solutions Setup Tool.
2) For any audio or text sent to a group.
3) To locate unknown channel dial code destinations. The VECPU6, VECPU6-EXP, VENSCA, VE8006R, and any VoIP endpoints using DHCP find unknown channel dial code destinations via **multicast**. These dial code routes are dynamically added to the phone book as found. If an entry in the phone book fails, perhaps due to the DHCP assigned address changing, then a new multicast query is sent to obtain current information and the phone book is updated. *If the unknown channel dial code cannot be located via multicast, users may receive a busy tone, an incomplete call message or call button generated calls may be unsuccessful.*

Regardless of the use of DHCP or static addressing, the use of groups, such as all call, require properly configured multicast.

If multicast is not properly implemented when using the VECPU6, VECPU6-EXP, VENSCA, VE8006R, and/or any VoIP endpoints using DHCP, there may be issues with calling both individual VoIP gateway and speaker channels and/or accessing groups.

A proper installation includes maintenance of multicast forwarding tables using IGMP querying (preferred), IGMP snooping, a combination of the two, or similar functions.

On a properly configured network, Valcom endpoints join the multicast group on start up. After that, whether they continue to receive multicast traffic is completely dependent upon network design. Enabling IGMP querying or equal is typically a good solution as the network routers will poll the devices to inquire as to whether they should remain in the multicast group. Valcom IP speakers and gateways will respond to these queries.

Other network methods of determining multicast group membership, such as IGMP snooping, may not be as effective since Valcom IP speakers and gateways only use multicast when required and may appear idle to IGMP snooping and subsequently be removed from the IGMP group. Note that if IGMP snooping and IGMP querying can both be implemented it is generally advisable to do so.

Networking Facts and Myths

VoIP based audio equipment may seem difficult for those unfamiliar with computer networking. There are many requirements for ports, protocols and routing that may seem intimidating.

Most of these requirements pertain to installing VoIP devices on a shared network.

If your application does not require routing to the Internet, a non-Valcom NTP server, a mail server, or other support resources, then a simple solution may be connecting the VoIP endpoints to dedicated, unmanaged PoE or PoE+ network switches. This will result in full compliance with the network requirements without any network configuration at all. Interconnected, unmanaged switches will, by nature, not block any ports or protocols.

Many integrators will preprogram and verify the operation of VoIP systems using unmanaged switches. Using this method, if there is a communication problem when the VoIP devices are deployed on the customer's network, then the issue is obviously in that network's configuration.

VLAN
All endpoints on a data network must, at a minimum, analyze each packet of data to determine if it is relevant information.

Imagine if the Postal Service didn't sort your mail and simply showed you every envelope and package. You would have to look at each one to determine if it was your mail. All of this "busy work" would keep you from doing what you need to do. Fortunately, the Postal Service does sort the mail and delivers it to the intended recipient.

Likewise, most networks have routers or managed switches that route traffic to the correct LAN segment, however, LAN segments typically have more than one host, so there is still irrelevant traffic to address. Most endpoints will exhibit slowed response during high traffic periods. This is why endpoints responsible for replicating real time information, like video or voice, should be on a VLAN. If VoIP devices are busy addressing excessive irrelevant network traffic, the quality of the audio will suffer.

Multicast
There are many myths about multicast and network security issues. Multicast is no less secure than unicast. Network endpoints that use multicast must request to join the multicast group; those that do not use multicast simply ignore the packets. In a properly configured network, with maintained multicast forwarding tables, multicast traffic is routed only where necessary. The use of multicast:

a) Saves significant bandwidth when sending the same audio/text to multiple endpoints.
b) Provides for audio synchronization between multiple audio endpoints.

Network Terms

OSI Model

The Open Systems Interconnection (OSI) model is a tool for understanding data communications between networked endpoints/systems. Each of the 7 layers provides services to support the layers above it and offers services to the layers below it.

The first three layers, the lower layers, deal with the details of how information is sent between networked computers. Layers 4–7, the upper layers, deal with how computer application programming interfaces communicate through the network.

LAYER 1: THE PHYSICAL LAYER

The bottom layer of the OSI Model is the Physical Layer.

- A. physical characteristics of the network
     - a. types and length of cables used to connect devices
     - b. types of connectors used, how long the cables can be, and so on.
- B. Electrical characteristics of the data signals
     - a. 0 and 1
          - i. No actual information protocols are defined
- C. Sample Layer 1 devices
     - a. Repeaters
     - b. Hubs
          - i. No routing here

LAYER 2: THE DATA LINK LAYER

- A. Significance (meaning) of transmitted bits
- B. Addresses
     - a. Size of each packet of data to be sent
     - b. How packets are addressed for proper
     - c. Coordination of data transmission from multiple nodes
     - d. Basic error detection and management
- C. MAC addresses are used to identify and direct traffic to hosts
- D. Sample Layer 2 devices
     - a. Bridges
     - b. Modems
     - c. Network cards
     - d. Layer 2 switches

LAYER 3: THE NETWORK LAYER

 A. Routing network traffic/messages from one computer to another
  a. Internet Protocol (IP) is an example Layer-3 protocol
 B. Manages logical addressing
  a. IP addresses are an example
   1. The Network Layer translates logical addresses to MAC addresses. This is necessary because the IP protocol uses a Data Link Layer protocol to transport packets to devices, so IP must know how to correlate the IP address and MAC address of a device.
 C. Manages routing between computers on different networks.
  a. Can route between dissimilar networks (T1, Cable, Ethernet, wireless, etc.)
 D. Sample Layer 3 devices
  a. Routers
  b. Layer 3 switches

LAYER 4: THE TRANSPORT LAYER

 A) The Transport Layer is the layer at which network computers communicate with other network computers.
 B) Transmission Control Protocol (TCP) lives here.
  a. ensures that packets move over the network reliably and error free.
   i. establishes connections between network devices
   ii. acknowledges the receipt of packets
   iii. resends packets as necessary
   iv. decompresses compressed data
  b. Transport Layer protocol may divide large packets into smaller packets for efficient transmission. Reassemble them as the receiver and verify the integrity of the information that was sent.

LAYER 5: THE SESSION LAYER

 A) Establishes and properly maintains communication between network nodes. This is necessary before data can be transmitted over the network.
  a. Handles authentication
  b. Handles authorization

LAYER 6: THE PRESENTATION LAYER

 A) Responsible for how data is represented on the network
  a. Converts data from one form to another form
   i. Data compression
   ii. Data conversion
   iii. Data encrypting
   iv. Data unencrypting

LAYER 7: THE APPLICATION LAYER

   A. Represents the methodologies/techniques that application programs (such as Spreadsheet, Word processor applications or Web browsers) use to communicate with the network.
   B. Examples:
      a. HTTP resides here
      b. Remote login to hosts (Telnet)
      c. File Transfer Protocol (FTP)
      d. Simple Mail Transfer Protocol (SMTP)
      e. Domain Name System (DNS)
      f. Simple Network Management Protocol (SNMP)

Layer 2 vs. Layer 3 Switches

By default, all the switch ports on a layer 2 switch are in the same VLAN and therefore share all broadcast traffic. Layer 2 switching is accomplished by the switch learning which unique MAC address is associated with each port. It then uses this information to deliver packets to their intended destination. The switch learns the MAC address/port mapping by monitoring, and learning from, host traffic. When a layer 2 switch encounters a previously unknown MAC address, all ports are flooded to locate the new host/endpoint.

Layer 2 switch ports can be assigned to different VLANs which results in multiple broadcast domains and helps to prevent broadcast storms. Layer 2 switches cannot provide routing between these different VLANs. Instead, routers or layer 3 switches are used for intercommunication between these VLANs.

For example, suppose we have made 2 VLANs, one for Valcom IP endpoints (VLAN 1) and one for another network traffic (VLAN 2). Let's assume that the site's NTP server is on VLAN 2. A router or layer 3 switch may be uses to allow the Valcom endpoints to access the NTP server.

Layer 3 switches are multi-layer switches and, in addition routing between VLANs can provide all layer 2 switch functions.

So, what's the difference between Layer 3 switches and routers? Layer 3 switches differ from routers in that they provide faster routing. In general, a layer 3 switch connects hosts to form local area networks, they do not have WAN port(s). Routers on the other hand, connect multiple local area networks to form wide area networks. Additionally, routers typically have more advanced capabilities than layer 3 switches. Routers use the Layer 3 source and destination IP addresses to determine the correct forwarding path for received packets.

[Spanning Tree](#)

The Spanning Tree Protocol is a Layer 2 protocol. The spanning tree algorithm is used in switch-based networks to determine the best path for traffic to move between two network devices. You can think of spanning tree as a navigation system for network traffic.

Spanning tree algorithm prevents [loops](#) and promotes efficient data delivery by discovering all possible paths between two ports on a network and ensures that only one is active at a time.

If the primary path is not available, then spanning tree protocol directs traffic to alternate paths if available.

Spanning tree protocol utilizes a primary decision-making switch called the root switch. All switches in a network utilizing spanning tree protocol branch from the root switch and all network paths are traceable back to the root switch. The root switch is necessary as it serves as a reference point for the best route calculations. The root switch is analogous to a navigation system's starting point.

Data messages exchanged between switches, known as bridge protocol data units (BPDUs), gather information about other switches in the same network.

Each port on a switch using spanning tree protocol may be in any of five states – blocking, listening, learning, forwarding or disabled.

With spanning tree enabled, every switch port in the network starts in the *blocking* state, then moves to the *listening* and *learning* states. If a potential loop is detected, the port will be set to the *blocking* state. If no loop is detected, the port will be set to the *forwarding* state. The ports then sustain the forwarding or blocking state unless changes are made to the network.

The bridge protocol data unit exchange is used to determine which network switch will serve as the root switch.

A switch always enters the blocking state following switch initialization and will not send or receive any data traffic across the network segment. A port in the blocking state will however listen to BPDU messages.

If you wish to force a port into forwarding state when connected, bypassing the blocking, listening, and learning states, then program the port in Edge Port or equivalent mode.

[PIM](#) (Protocol Independent Multicast)

Protocol Independent Multicast (PIM) is a layer 3 Multicast control protocol. PIM advertises multicast sources and receivers over a routed layer 3 network.

There are three PIM protocols, PIM Sparse Mode, PIM Dense Mode and, the less often used Bi-directional PIM.

PIM Sparse Mode protocol assumes that most subnets in the network will not require Multicast traffic. To receive Multicast data relevant to certain sources and groups, routers must request it from their upstream neighbors. Routers use PIM Join and Prune messages to join and leave Multicast distribution trees. Multicast distribution trees are rooted at some selected router (i.e. Rendezvous Point or RP). This RP is used by all sources sending to the Multicast group. One of the important requirements of PIM Sparse Mode is the ability to use a shared tree to discover the address of the RP for a Multicast group.

With PIM Sparse mode, the switches keep track of the ports that want multicast by detecting IGMP Query responses sent to the ports by some Querier device. That information gets forwarded upstream toward the source.

PIM Sparse Mode uses unicast routing information to create a distribution tree along the reverse path (receivers to source). This is known as Reverse Path Forwarding (RPF). In the process PIM learns of both upstream and downstream neighbor hosts. With PIM, a router will forward a Multicast packet only if it is received from the upstream neighbor. This helps to ensure that the distribution tree will be free of loops.

If the source and group information isn't kept refreshed, then multicast can stop being delivered.

Packet captures of the switch port where misbehaving multicast dependent endpoints are connected can provide a lot of information. Generally, for IGMP troubleshooting, the captures need to run for 4 or 5 minutes to get enough data. For IP Speaker one way audio failures, a quick way to see if IGMP could be the problem is to wait for the speaker to fail to play a page. Then reboot the speaker and try a test page. If it plays correctly, wait for at least 10 minutes and try another test page. If that fails, then the multicast routing entries could be timing out and getting dropped from the routers/switches. And if you reboot the speaker and it starts playing pages again, then it strongly suggests a problem with IGMP.

With PIM Dense Mode (PIM-DM), a second routing protocol, Multicast data is initially sent to all hosts in the network. If none of the hosts on a router are interested in receiving Multicast, then the routers send PIM Prune messages to remove themselves from the tree.

Eventually, the Prune state at each router times out, and data will begin to flow back into the parts of the network that were previously pruned. If the hosts on any router still lack interest in the Multicast traffic, further Prune messages will be sent.

IGMP (Internet Group Management Protocol)

IGMP is a Layer 3 communications protocol used by hosts and adjoining routers to establish and maintain Multicast group memberships on IPv4 networks. This is usually accomplished by use of IGMP querying, the more passive IGMP snooping or a combination of the two. Multicast enabled endpoints that are prone to idle states where they will not receive Multicast are best served by the proactive qualities of Multicast Querying.

IPv4 vs IPv6

IP addresses identify hosts on an addressable data network. IPv4 supports 32-bit IP addresses while IPv6 supports 128-bit IP addresses. There are not enough IPv4 addresses for every network device worldwide, therefore, network address translation (NAT) is used to allow local networks to all use the same addresses. Traffic leaving these local networks uses a unique shared IPv4 address. NAT manages the private to public network routing. IPv6, once fully implemented, will allow for a unique IP address for every network device thereby eliminating the need for NAT.

Network Interface Controller (NIC)

A network controller implements the electronic circuitry required to communicate using a specific physical layer (cable, fiber) and data link layer standard such as Wi-fi or Ethernet.

Wide Area Network (WAN)

A computer network spanning remote locations such as regions, cities, countries.

Local Area Network (LAN)

A computer network that interconnects facilities within a limited area, such as one or more buildings. Multiple LANs may be interconnected with a Wide area Network (WAN)

Virtual Lan (VLAN)

A Virtual Network (VLAN) Partitioned and isolated bandwidth use to isolate related network endpoints from general network traffic. Analogous to an HOV highway lane. A network within a network.

Virtual Private Network (VPN)

A Virtual Private Network (VPN) is used to extend a private network to a remote host by use of public bandwidth (i.e. The Internet). A VPN uses encryption to ensure data privacy.

[Bandwidth](#)

Bandwidth is the maximum rate of transfer across a network. Bandwidth is adversely affected by the amount of traffic on a network.

[Unicast](#)

Unicast is one sender to one receiver network communication.

[Multicast](#)

Multicast is a network protocol where one sender relays the same information to multiple receipts simultaneously. Multicast is typically used for bandwidth conservation and synchronization of recipient endpoints.

Know Your Ports

When integrating internal communication systems to a facility telephone system, several common access methods are used.

Loop Start Trunk Port Access – If you plan to connect to one or more telephone system loop start trunk ports, you will, at a minimum, need a device that provides battery feed voltage. If a device provides battery feed, you will measure dc voltage across its connection pair. The voltage is typically -24 or -48 vdc.

If the device that you are connecting to the loop start trunk port has multiple zones that are dial selectable, then you will also be able to place your lineman's handset across its connection pair, go off-hook and hear some form of dial tone. Dialing a DTMF digit should "break" the dial tone.

If the device that you are connecting to the loop start trunk port features call switches, then you will also be able to place your on-hook lineman's handset across its connection pair, press a properly configured call switch and ring your lineman's handset.

Loop start trunk ports are also known as FXO ports and Loop Start C.O. Line Ports.

Analog Station Port Access – If you plan to connect to one or more telephone system analog station ports, you will need a device that can automatically answer from ring voltage and that can automatically disconnect once the call is terminated. Possible disconnects, in the order of preference are:

a) Open loop disconnect detection (ideal)
b) Silence detection
c) Absolute timeout

If the device that you are connecting to features call switches, then connecting to an analog station port, even if feasible will require additional engineering.

Analog station ports are also known as FXS ports.

> *If you measure dc voltage across an unterminated port on a telephone system, it may be an analog station port, but it's __not__ a page port or loop start trunk port!*

Page Port Access – Page ports on telephone systems typically provide a line level audio output and a contact closure. In most cases they do not provide zone selection capability and should only be used for single zone systems.

Session Initiation Protocol (SIP) Access – SIP access is easily accomplished, not only to SIP ready devices, but to any device via the proper SIP gateway.

Signal Attenuators

Different signal attenuators are available for different applications. Which one should you use?

V-LPT Impedance Matching Module
The V-LPT is designed to connect to 45 Ohm system speaker outputs. It attenuates the speaker signal level (≈ 7vrms) to line level audio (≈ 245mv rms) and lowers the output impedance to 8 Ohms. It can provide one-way audio to up to 150 Valcom Self Amplified Speakers or can feed line level audio into an old fashioned 25/70.7/100-volt amplifier. Although it includes a trim pot for fine tuning the signal, it is really designed to be setup during installation and is not for daily adjustment.

V-1092 Wall Mount Volume Control
The V-1092 is a single gang passive (non-powered) volume control for line level audio sources. It is designed to be adjusted as often as necessary. It is a very useful addition to speaker lines serving speakers or horns that will be difficult to access after installation, speakers in individual offices or meeting rooms, or zones consisting of multiple speakers in a common area like a cafeteria or hallway. The V-1092 features a stainless-steel plate and fits a single gang electrical wall box.

V-1094A Page Port Preamp/expander
The V-1094A is a single gang active (powered) volume control/preamp for line level audio sources. Although not typical, it is designed to be adjusted often if necessary. It boosts signal on line level audio lines reaching capacity and provides an 8-ohm output to drive up to 150 additional Valcom Self Amplified Speakers or horns. The V-1094A features a stainless-steel plate and fits a single gang electrical wall box.

V-1095 70 Volt Expander
The V-1095 may be used to attenuate the audio output of an old fashioned 25/70.7/100-volt amplifier or 25-volt intercom station card outputs to line level audio. The V-1095 features a stainless-steel plate and fits a single gang electrical wall box. It also features a volume control for fine tuning the signal.

VMT-1 Impedance Matching Transformer
The VMT-1 provides an impedance match between a 500-600 Ohm balanced line and a high impedance auxiliary or microphone input (typically of an old fashioned 25/70.7/100-volt amplifier). It may also be used to attenuate the audio output of an old fashioned 25-volt amplifier to line level audio (Although the V-1095 is preferred for this application).

Additionally, a VMT-1 may be used to transform -10dBm at low impedance to @ 1 V rms at 10k if needed (turns ratio is about 1:3.89 as a step up).

V-1096 Audio Interface
The V-1096 provides:

a) line level audio to Lo-Z microphone level audio
b) line level audio with a normally open contact closure to FXS (station port)
c) DC decoupling with a 4.7K supervision resistor (input) in all cases

S-560 20-Watt Attenuator
The S-560 permits changes to the audio level of 25- or 70.7-volt speakers and horns without altering the speaker wiring connections. Tap the speaker for the maximum audio level required and use the attenuator to change the volume level. The attenuator features a stainless-steel plate with embossed numbers and fits a single gang electrical wall box. The attenuator controls a 20-watt speaker load with a 10-position tap switch and operates on 25- or 70.7-volt speaker lines.

The S-560 is also the attenuator of choice for 45-ohm talkback speaker circuits. Red/Black in Green/Black Out.

Call switches with volume control
There are several models of call switches for 45-ohm talkback speaker circuits that feature integrated volume controls.

V-2971 Normal call switch with volume control (Stainless Steel)
V-2991-W Normal call switch with volume control (White Decor)
V-2970 Normal/Emergency call switch with volume control (Stainless Steel)
V-2995-W Normal/Emergency call switch with volume control (White Decor)

These are designed to provide convenient screwdriver adjustment of speaker broadcast levels.

When these are used, both the audio and call switch pairs of the originating equipment terminate on the call switch with volume control. A new attenuated pair is then routed from the call switch/attenuator to the speaker. For maximum range, adjust the 45-ohm speakers being controlled to their highest volume level.

IP Addresses, Netmasks, Subnet Masks and Network Class

Even if you're not managing the IP network, it's helpful to understand a bit about the information that will be provided. Blindly accepting an invalid range of IP addresses and a subnet mask can cause major site delays and installation costs.

Subnets are divisions of larger networks.

A subnet mask is a 32-bit binary number that divides an IP address into network address and host address. A subnet mask is made by setting network bits to all "1"s and setting host bits to all "0"s. We often view it in its binary equivalent form.

For example, 11111111 11111111 11111111 00000000 = 255.255.255.0

How does it work?

You have an IP address of 128.42.5.4 and a netmask of 255.255.248.0

IP address 128.42.5.4 = 10000000 00101010 00000101 00000100 in binary

Netmask 255.255.248.0 = 11111111 11111111 11111000 00000000 in binary

The *prefix* of 128.42.5.4 with a 255.255.248.0 netmask is /21 because there are 21 contiguous "1" s in the most significant (leftmost) part of the Subnet mask. Therefore, we can write our IP address as 128.42.5.4/21

In order to determine the part of the IP address that indicates the network of which it is a part (the network address or netmask), perform a logical conjunction of the binary representation of the IP address and the netmask:

10000000 00101010 00000101 00000100 11111111 11111111 11111000 00000000 10000000 00101010 00000000 00000000	Logical Conjunction In easy terms, if both numbers in any position are "1" then the result in the network address is "1", otherwise it's "0".

So, the network address of all IP addresses in 128.42.5.4/21 is:

10000000 00101010 00000000 00000000 or 128.42.0.0

There are 32 digits in the binary representation of an IP address. In our case the subnet mask indicated that the first 21 represent the network address, therefore the last 11 are allocated to represent host addresses – from 000 00000001 to 111 11111110.

The host address that is all "1" s will be the broadcast address to which all hosts on this subnet will listen.

The host address that is all "0" s will be the network address.

Both the network address and the broadcast address are reserved and cannot (by current and previous network standards) be assigned to a device.

Therefore, our useable range of host addresses on this subnet are:

HostMin: 128.42.0.1   10000000 00101010 00000**000 00000001** (in binary)
HostMax: 128.42.7.254   10000000 00101010 00000**111 11111110** (in binary)

There are 2046 hosts available on this subnet

128.42.0.1 – 128.42.0.255 = 255

128.42.1.0 – 128.42.1.255 = 256

128.42.2.0 – 128.42.2.255 = 256

128.42.3.0 – 128.42.3.255 = 256

128.42.4.0 – 128.42.4.255 = 256

128.42.5.0 – 128.42.5.255 = 256

128.42.6.0 – 128.42.6.255 = 256

128.42.7.0 – 128.42.7.254 = 255

(256 x 6) + (255 x 2) = 2046 host addresses

The broadcast address is the host address that is all "1" s or:

Broadcast: 128.42.7.255   10000000 00101010 00000**111 11111111** (in binary)

An IPv4 Address – Dotted Decimal Notation

128 . 42 . 5 . 1

10000000 00101010 00000101 00000001

One Byte   =   8 Bits

Thirty-two bits (4 x 8), or 4 bytes
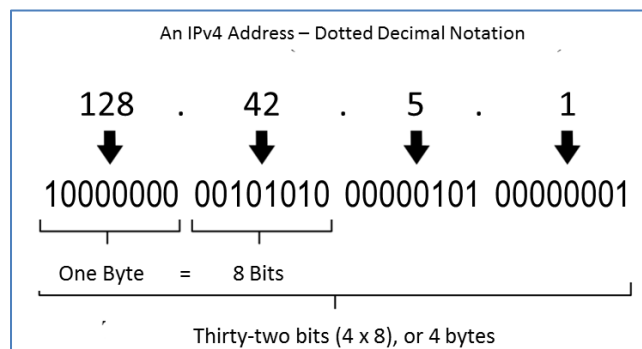
Divide 128.42.0.0/21 into 4 equally large subnets of at least 100 hosts each

To do this we will use some of the host bits in 128.42.0.0/21 to designate smaller subnets.

As an exercise, we will break 128.42.0.0/21 into 4 subnets that must hold at least 100 hosts each...

How many least significant (leftmost) host bits will be required to create a subnet of at least 100 hosts?

$$
\begin{aligned}
1 &= 1 \\
11 &= 3 \\
111 &= 7 \\
1111 &= 15 \\
11111 &= 31 \\
111111 &= 63 \\
1111111 &= 127
\end{aligned}
$$

So, it will take 7 bits to have at least 100 host addresses.

Another way to calculate this is with the formula:

Host bits = Log_2 (# Hosts Desired) = Log_2 (100) = 6.643 ≈ 7

Since IPv4 addresses are 32 bits wide[1], and we are using least significant host bits to address our 127 hosts, simply subtract 7 bits from 32 bits to calculate the minimum subnet prefix for each subnet... 32 - 7 = 25.

But we'll use /24 because it falls on an octet boundary (3 full octets) and will provide more than the 100 host addresses that we desire.

32 total bits – 24 bits = 8 bits available for our host IP addresses.

We are going to use the last 8 bits of our IP addresses for host addresses

10000000 00101010 00000000 *hhhhhhhh*

hhhhhhhh will range from 00000001 to 11111110

Therefore, there are 24 (32 – 8) bits available to represent the subnets.

We only need 4 subnets so we will use the lowest 4 subnet addresses available.

000 = 0
001 = 1
010 = 2
011 = 3

Note that there are actually 8 potential subnets available with the 3 bits we gained from the original host addresses (going from a /21 to a /24 mask).

128.42.0.0

Network Address	Host Number		128.42.0.0/21
Network Address	Subnet 1	Host Number	128.42.0.0/24
Network Address	Subnet 2	Host Number	128.42.1.0/24
Network Address	Subnet 3	Host Number	128.42.2.0/24
Network Address	Subnet 4	Host Number	128.42.3.0/24

The first 3 octets of the new *subnet* mask will all be "1" s (8 bits per octet x 3 octets = 24). In binary 11111111 11111111 11111111 00000000 or in decimal 255.255.255.0. Applying this *subnet* mask to any IP address in the subnet will return the subnet address.

For example:

IP address128.42.3.55 decimal is 10000000 00101010 00000011 00110111

With subnet mask 255.255.255.0 which has a decimal equivalent of 11111111 11111111 1111111 00000000

10000000 00101010 00000011 00110111 = 128.42.3.55

11111111 11111111 11111111 00000000 = 255.255.255.0 (subnet mask)

10000000 00101010 00000011 00000000 = 128.42.3.0 subnet address

Applying this *netmask* to any IP address will return the network address.

10000000 00101010 00000011 00110111 = 128.42.3.55

11111111 11111111 11111000 00000000 = 255.255.248.0 (netmask)

10000000 00101010 00000000 00000000 = 128.42.0.0 network address

Subnet masks applied to IP addresses reveal the subnet address.

Netmasks applied to IP addresses reveal the unsegmented network address.

The host addresses that are all "1" s will be the broadcast addresses to which all hosts on this subnet will listen.

128.42.0.255 = 10000000 00101010 00000000 **11111111**

128.42.1.255 = 10000000 00101010 00000001 **11111111**

128.42.2.255 = 10000000 00101010 00000010 **11111111**

128.42.3.255 = 10000000 00101010 00000011 **11111111**

The host address that is all "0" s will be the subnet address.

128.42.0.0 = 10000000 00101010 00000000 **00000000**

128.42.1.0 = 10000000 00101010 00000001 **00000000**

128.42.2.0 = 10000000 00101010 00000010 **00000000**

128.42.3.0 = 10000000 00101010 00000011 **00000000**

Both the subnet address and the broadcast address are reserved and cannot (by current and previous network standards) be assigned to a device.

Therefore, our useable range of host addresses on these subnets are:

On subnet 128.42.0.X:

HostMin: 128.42.0.1   10000000 00101010 00000000 **00000001** (in binary)
HostMax: 128.42.0.254   10000000 00101010 00000000 **11111110** (in binary)

There are 254 hosts available on this subnet 128.42.0.1 – 128.42.0.254 = 254

On subnet 128.42.1.X:

HostMin: 128.42.1.1   10000000 00101010 00000001 **00000001** (in binary)
HostMax: 128.42.1.254   10000000 00101010 00000001 **11111110** (in binary)

There are 254 hosts available on this subnet 128.42.1.1 – 128.42.1.254 = 254

On subnet 128.42.2.X:

HostMin: 128.42.2.1   10000000 00101010 00000010 **00000001** (in binary)
HostMax: 128.42.2.254   10000000 00101010 00000010 **11111110** (in binary)

There are 254 hosts available on this subnet 128.42.2.1 – 128.42.2.254 = 254

On subnet 128.42.3.X:

HostMin: 128.42.3.1   10000000 00101010 00000011 **00000001** (in binary)
HostMax: 128.42.3.254   10000000 00101010 00000011 **11111110** (in binary)

There are 254 hosts available on this subnet 128.42.3.1 – 128.42.3.254 = 254

[1]32 bits is 8 bytes xxxxxxxx.xxxxxxxx.xxxxxxxx.xxxxxxxx

Network Class

IP Address Class	Total # Of Bits for Network ID / Host ID	First Octet of IP Address	Usable # Of Network ID Bits	# of Possible Network IDs	# Of Host IDs Per Network ID
Class A	8 / 24	**0**xxx xxxx	8 - 1 = 7	$2^7 - 2 = 126$	$2^{24} - 2 = 16,277,214$
Class B	16 / 16	**10**xx xxxx	16 - 2 = 14	$2^{14} = 16,384$	$2^{16} - 2 = 65,534$
Class C	24 / 8	**110**x xxxx	24 - 3 = 21	$2^{21} = 2,097,152$	$2^8 - 2 = 254$

First Octet for Class A = 1 – 127 = **0**000 0001 – **0**111 1111

First Octet for Class B = 128 – 191 = **10**00 0000 – **10**11 1111

First Octet for Class C = 192 – 223 = **110**0 0000 – **1101** 1111

To find the number of possible network IDs or host IDs, use the formula:

$$2^{\text{USEABLE NUMBER OF ID BITS}}$$

Refer to the last 2 columns in the chart shown above for examples.

There are many online subnet calculators like the ones found here:

http://www.subnet-calculator.com

http://jodies.de/ipcalc (Here is the result of our examples using ipcalc)

Use Log Files as a Tool

Most network based Valcom equipment has the ability to send information about activity to a syslog server on port 514. Often, there are optional levels of syslog reporting available offering minimal to maximum data.

There are many ways to store syslog data. The Valcom IP Solutions Setup Tool and most other Valcom programming/setup tools offer a built-in syslog viewer and feature the ability to log the data to a file on the PC.

Some servers can use the content of syslog data to trigger messages via e-mail to alert users of system activity or malfunction. VE6024 eLaunch servers have a log file in their zipped server backup file.

In addition, Valcom Application servers and telephone paging servers all feature a log file in their browser interfaces. Most Valcom IP endpoints feature a console output that can be viewed with the VIP Utility Tool. These log files and console outputs provide real-time information about device input and output activity.

Syslog can also be very beneficial for troubleshooting systems by documenting events leading up to, during, and following troubles.

Assume Nothing.

Question Everything.

Understanding Ground Loops

In an ideal world, measurements between various connections to Earth ground would vary by zero volts worldwide. Unfortunately, this is not the case.

Ground loops are a major cause of noise and hum in audio and other electronic systems.

Ground loops occur when two or more stages of an audio circuit are referenced to two or more "ground" points of different electrical potential. In other words, you would measure voltage between the two grounds with a voltmeter. The difference in potential between the ground points allows current to flow between ground 1 and ground 2, which induces noise across the connection path. It's common for different ground points within a facility to exhibit potential difference with respect to each other.
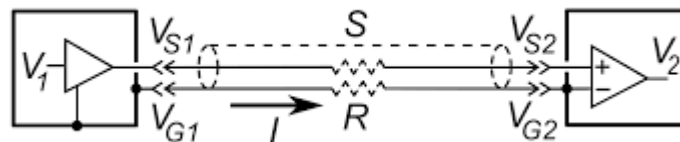
In the diagram above, the signal V_1 is applied to the conductor of signal cable S by an unbalanced line driver in component 1. The cable has a ground conductor which is attached to the metal chassis of each component. At the destination end in component 2, the signal and ground are connected to a differential amplifier to produce the signal V_2. If the chassis of the two components are at different potentials, a current "I", will flow through the ground conductor, which has a resistance R, causing a voltage drop IR along the cable ground. This voltage will be added to the signal voltage by the differential amplifier in component 2, resulting in interference on the signal.

If current "*I*" is an AC current, and the system is an audio system, noise or hum may be heard in the speakers. This occurs because the difference in potential between the ground references is summed with the received signal. (i.e. if the relative reference potential moves up 0.1v then the desired signal will move up 0.1v as well) So, as a function of time, all of the variances in the receiver's ground reference are added to or subtracted from the desired signal. This results in audible noise.

Preventative steps

Whenever practical, ensuring that all vulnerable signal circuits are referenced to one ground point via a very low resistance path is the best way to avoid ground loops. Other commonly deployed techniques include:

1. Using differential drivers, receivers and UTP cabling to reject ground-induced interference.

2. Removing ground connections to equipment. Note that this may cause safety issues and may not be permitted by the relevant electrical code.

3. Transformer coupling audio between circuits stages having different ground points. The transformers allow the audio to pass but remove any ground reference. Valcom VMT-2s are typically used for coupling line level audio in an effort to eliminate ground loops.

Advantages of Unshielded Twisted Pair Cable
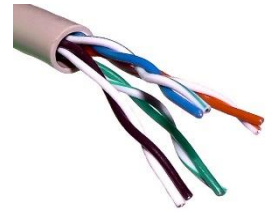
Valcom capitalizes of the use of unshielded twisted pair cable. Highly reliable UTP was developed for the telephone and networking industries. There are many advanced cable management and connectivity options for UTP such as 110 and 66 blocks. In addition, many cable manufacturers offer long term warranties on their UTP products. UTP, when used with differential signals and inputs provide inherent noise rejection and help to minimize the effects of capacitance between adjacent pairs.

To understand the magic of unshielded using twisted pair (UTP) wiring, one must be familiar with a few concepts:

1. An electrical conductor, such as wire, will be surrounded by an electromagnetic field, the strength and direction of which is a function of the current flowing through the conductor.
    a. This means that, any wire carrying AC signal, such as audio, is surrounded by a magnetic field, the intensity and direction of which mimics the AC signal.
2. Secondly, an electrical conductor in a circuit, such as wire, when exposed to a changing magnetic field, will exhibit an induced electrical current that is representative of that changing magnetic field. The more exposure the conductor has to the changing magnetic field, the stronger the coupled signal.
    a. This means that any wire in a circuit that is routed in parallel to an AC current carrying conductor, as described in Item 1, will "pickup" the signal from that conductor via electromagnetic coupling. The longer the parallel routing, the more signal is "picked up". This is commonly called "crosstalk".

So, when we route cables together, how do we keep the AC signals from inadvertently coupling into each other? There are several techniques, but this discussion centers around unshielded twisted pair cable. Two more things to understand:

3. The potential difference (a.k.a. voltage) between 2 points exhibiting the same voltage will be zero.
    a. The difference between two like things is zero (+5V and +5V, -5V and -5V, etc.)
4. The potential difference between 2 points exhibiting 180° out-of-phase signal voltage will be doubled.
    a. i.e. the difference between -5V and +5V is 10V.
5. the dynamic forces of 2 magnetic fields occupying the same space will sum. If they are at opposite polarity the sum will be zero.

a. Any coexisting, coinciding forces that are of equal magnitude, but opposite direction will nullify each other. (i.e. 5 lbs. to the left applied to 5 lbs. to the right is zero force.)

So, what if we always sent the AC signal though both conductors in a wire pair, and we twisted these wires to ensure that they occupy the same physical space?

We'd have an electromagnetic force around each, right? Yes!

But if we sent the AC signal in one of the conductors 180° out-of-phase with the AC signal in the second conductor, the electromagnetic fields would also be 180° out-of-phase. This means that the fields would cancel each other out. There would be no EMF to induce undesired signal into surrounding conductors.
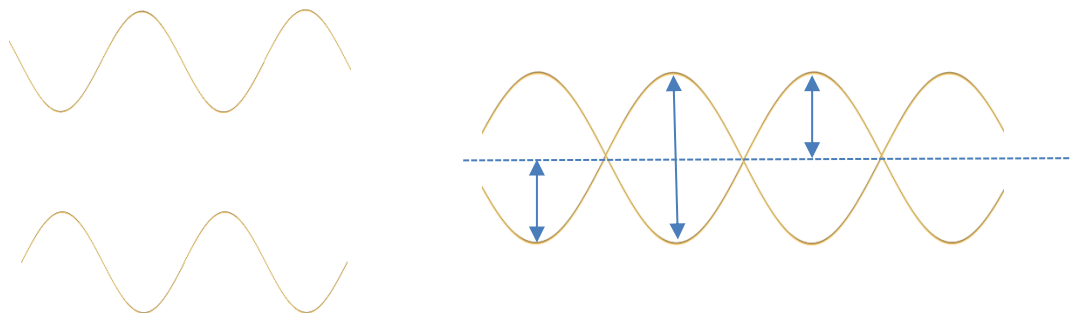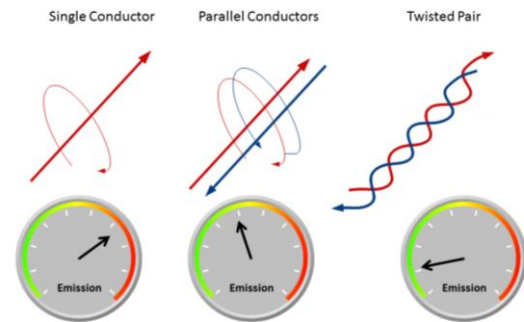
Out-of-phase Signals   Combined

The signal voltage between the 2 out-of-phase signals at any point is 2x the voltage of just one of the signals if it was referenced to the ground.

And because the UTP wire is twisted together and, as a result, virtually occupies the same physical space, if it happens to pick up electromagnetic interference from other sources, the interference will be induced on both wires identically; same magnitude, same direction.

When the wires connect to their destination (amplifier, handsfree audio receiver, etc.), any identical signal levels will result in zero signal (see item 3 above) across the receiver's input, this is referred to as common mode rejection.

Simultaneously, the 180° out-of-phase desired signal will be delivered to the receiving input at twice its amplitude (see item 4 above). Inputs used with UTP are called differential inputs.

Sample Values

Time (seconds)		Differential Signal (Signal Voltage)		Induced Common Mode Noise (Voltage)		Differential Signal Voltage Delivered to receiving input	
		Wire 1	Wire 2	Wire 1	Wire 2	Difference in Desired Signal	Difference in Common Mode Noise
T+	0.0001	-4	4	0.354	0.354	8v	0v
T+	0.0002	-3.7	3.7	0.548	0.548	7.4v	0v
T+	0.0003	-3.4	3.4	0.667	0.667	6.8v	0v
T+	0.0004	-3.1	3.1	0.699	0.699	6.2v	0v
T+	0.0005	-2.8	2.8	0.642	0.642	5.6v	0v
T+	0.0006	-2.5	2.5	0.501	0.501	5v	0v
T+	0.0007	-2.2	2.2	0.289	0.289	4.4v	0v
T+	0.0008	-1.9	1.9	0.023	0.023	3.8v	0v
T+	0.0009	-1.6	1.6	-0.271	-0.271	3.2v	0v
T+	0.001	-1.3	1.3	-0.567	-0.567	2.6v	0v
T+	0.0011	-1	1	-0.840	-0.840	2v	0v
T+	0.0012	-0.7	0.7	-1.065	-1.065	1.4v	0v
T+	0.0013	-0.4	0.4	-1.221	-1.221	0.8v	0v
T+	0.0014	-0.1	0.1	-1.295	-1.295	0.2v	0v
T+	0.0015	0.2	-0.2	-1.280	-1.280	0.4v	0v
T+	0.0016	0.5	-0.5	-1.178	-1.178	1v	0v
T+	0.0017	0.8	-0.8	-0.997	-0.997	1.6v	0v
T+	0.0018	1.1	-1.1	-0.754	-0.754	2.2v	0v
T+	0.0019	1.4	-1.4	-0.470	-0.470	2.8v	0v
T+	0.002	1.7	-1.7	-0.171	-0.171	3.4v	0v
T+	0.0021	2	-2	0.116	0.116	4v	0v
T+	0.0022	2.3	-2.3	0.366	0.366	4.6v	0v
T+	0.0023	2.6	-2.6	0.557	0.557	5.2v	0v
T+	0.0024	2.9	-2.9	0.671	0.671	5.8v	0v
T+	0.0025	3.2	-3.2	0.698	0.698	6.4v	0v
T+	0.0026	3.5	-3.5	0.636	0.636	7v	0v
T+	0.0027	3.8	-3.8	0.491	0.491	7.6v	0v
T+	0.0028	1.4	-1.4	-0.470	-0.470	2.8v	0v
T+	0.0029	1.6	-1.6	-0.271	-0.271	3.2v	0v
T+	0.003	1.8	-1.8	-0.073	-0.073	3.6v	0v
T+	0.0031	2	-2	0.116	0.116	4v	0v
T+	0.0032	2.2	-2.2	0.289	0.289	4.4v	0v
T+	0.0033	2.4	-2.4	0.437	0.437	4.8v	0v
T+	0.0034	2.6	-2.6	0.557	0.557	5.2v	0v
T+	0.0035	2.8	-2.8	0.642	0.642	5.6v	0v

Series and Parallel Speaker Calculations

Generally, amplifiers designed to drive voice coil speakers can drive an impedance >= their output impedance. If the speaker load impedance falls below the amplifier output impedance, then more signal is lost in the amplifier than is delivered to the speakers. Understanding the effective impedance of series and parallel speakers will prepare you for properly designing voice coil speaker loads. - Online Calculator
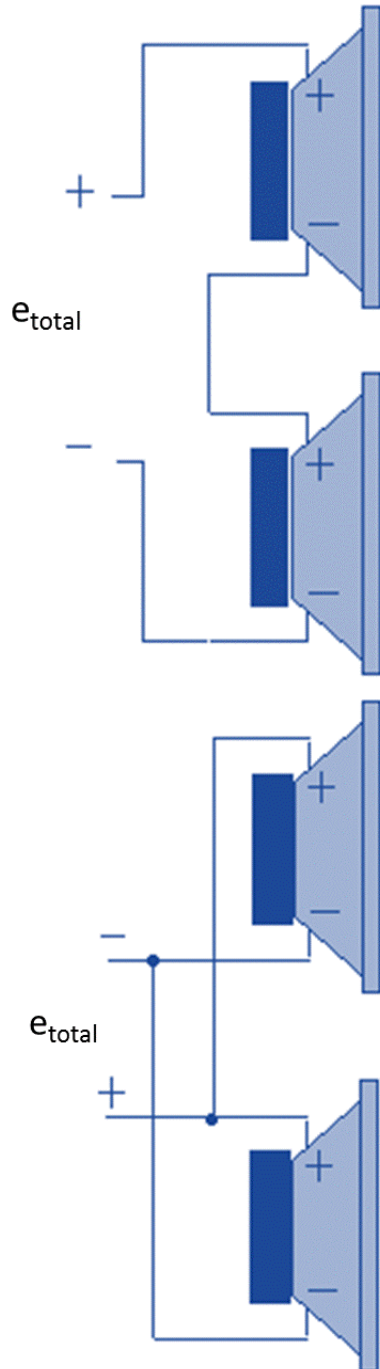
Speakers Connected in Series

z_t = total impedance of series speakers
z_{spkr} = impedance of any given speaker
e_{total} = source signal across series speakers
e_{spkr} = audio signal across any given speaker
p_{spkr} = power delivered to any given speaker

z_t = The sum of all speaker impedances

$$e_{spkr} = \frac{(e_{total})(z_{spkr})}{z_t}$$

$$p_{spkr} = \frac{e_{spkr}^2}{z_{spkr}}$$

Speakers Connected in Parallel

z_t = total impedance of parallel speakers
z_{spkr} = impedance of any given speaker
e_{total} = source signal across parallel speakers
e_{spkr} = audio signal across any given speaker
p_{spkr} = power delivered to any given speaker

$$z_t = 1/(\frac{1}{z_{spkr1}} + \frac{1}{z_{spkr2}})$$

$$e_{spkr} = e_{total}$$

$$p_{spkr} = \frac{e_{spkr}^2}{z_{spkr}}$$

Speakers Connected in Parallel 2

z_t = total impedance of parallel speakers
z_{spkr} = impedance of any given speaker
e_{total} = source signal across parallel speakers
e_{spkr} = audio signal across any given speaker
p_{spkr} = power delivered to any given speaker

$$z_t = 1/(\frac{1}{z_{spkr1}} + \frac{1}{z_{spkr2}} + \frac{1}{z_{spkr3}} + \frac{1}{z_{spkr4}})$$

$$e_{spkr} = e_{total}$$

$$p_{spkr} = \frac{e_{spkr}{}^2}{z_{spkr}}$$
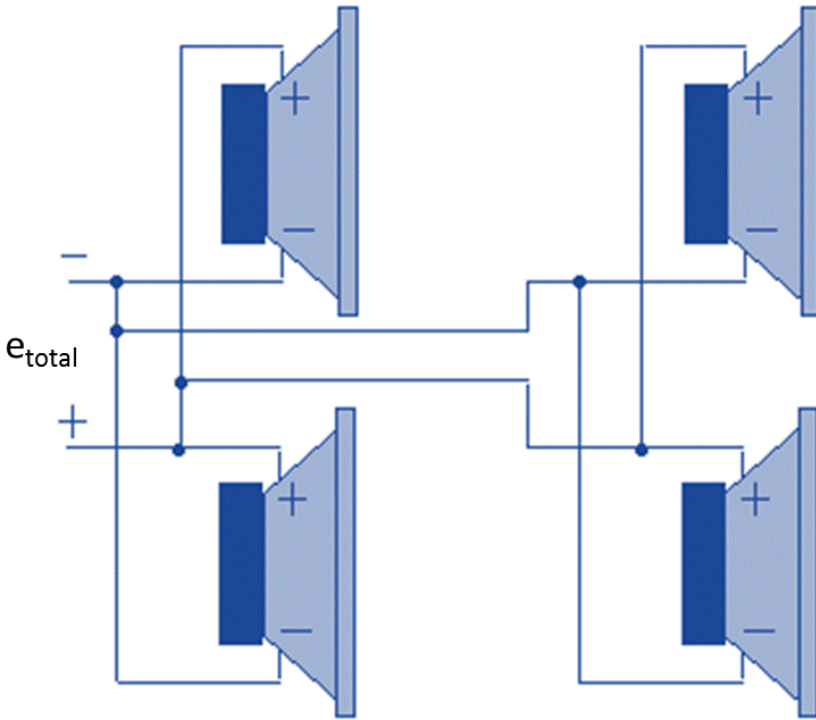
Speakers Connected in Series/Parallel

z_t = total impedance of parallel/series speakers
z_{spkr} = impedance of any given speaker
e_{total} = source signal across parallel/series speakers
e_{spkr} = audio signal across any given speaker
p_{spkr} = power delivered to any given speaker

$$z_t = 1/(\frac{1}{z_{spkra} + z_{spkrb}} + \frac{1}{z_{spkrc} + z_{spkrd}})$$

$$e_{spkrab} = e_{spkrcd} = e_{total}$$

$$e_{spkra} = \frac{(e_{total})(z_{spkra})}{z_{spkra} + z_{spkrb}}$$

$$e_{spkrd} = \frac{(e_{total})(z_{spkrd})}{z_{spkrc} + z_{spkrd}}$$

$$p_{spkr} = \frac{e_{spkr}{}^2}{z_{spkr}}$$
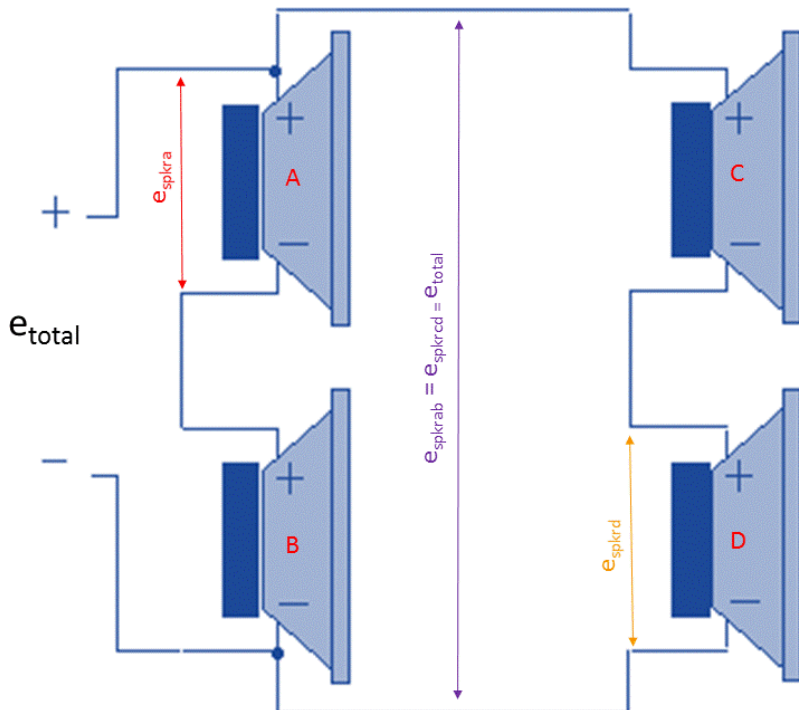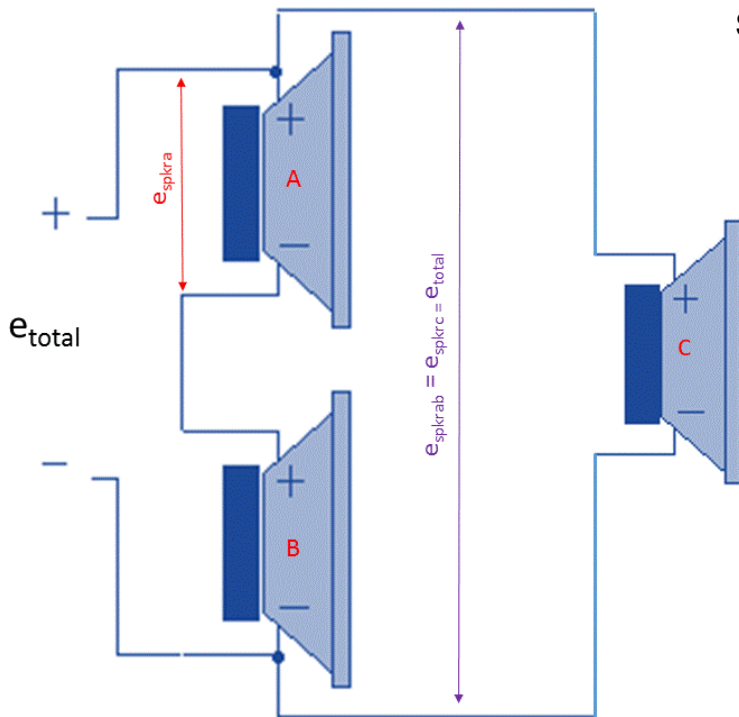
Speakers Connected in Series/Parallel 2

z_t = total impedance of parallel/series speakers
z_{spkr} = impedance of any given speaker
e_{total} = source signal across parallel/series speakers
e_{spkr} = audio signal across any given speaker
p_{spkr} = power delivered to any given speaker

$$z_t = 1/\left(\frac{1}{z_{spkra} + z_{spkrb}} + \frac{1}{z_{spkrc}}\right)$$

$$e_{spkrab} = e_{spkrc} = e_{total}$$

$$e_{spkra} = \frac{(e_{total})(z_{spkra})}{z_{spkra} + z_{spkrb}}$$

$$p_{spkr} = \frac{e_{spkr}{}^2}{z_{spkr}}$$

Negative Voltage

Valcom uses negative voltage (positive ground) for many of their products.

Equivalent negative and positive voltage have the same potential difference and the same capacity to perform work. Polarity is simply a matter of the ground or neutral reference point. If you measure a battery with a voltmeter and then swap the meter leads, you will observe a polarity reversal.

The reason that Valcom chose to use negative voltage dates to the infancy of the company when our marketing was targeted at the telephone company. The telephone industry standardized on -48vdc many years ago. Negative potential of telephone lines helps to prevent electrolysis related corrosion.

Valcom's roots in the telephone industry carry over into voltage polarity, the terms tip and ring for our audio pairs, the focus on the use of superior UTP cable and more. Incidentally, Valcom one-way self-amplified speakers are not polarity sensitive and will work with positive or negative voltage.

Electric Fields and Forces

Whenever electrical current flows through wire, there is a magnetic field (EMF) created around the wire.

If the wire carries dc current, then the magnetic field is steady state.

If the wire carries ac current, then the magnetic field will change polarity along with the current.

When multiple ac current carrying wires are routed in parallel, their magnetic fields will sum. If they are 180° out of phase the magnetic fields will sum to zero, or null. Otherwise the magnetic fields will enhance or deplete each other.

The magnitude of the combined fields depends upon the product of the charges and the square of the distance between the charges. In other words, separating the wires, diminished the combined EMF.

When surrounding wires are exposed to the changing magnetic field, the field induces signal into those wires.

Why do you need to know this?

Imagine that you have a site trouble where a certain amplifier is the broadcasting music/voice, when music/voice, is only sent to other areas?

The crosstalk issue could be caused by EMF.

The author recalls a jobsite where music was being broadcast to half of a facility, but it was heard in the whole facility.

Upon inspection, all facility audio was sent through the same 50 pair cable. The combined EMF of the in-phase music signal in ½ of the cable conductors was inducing music signal into the second ½ of the cable conductors. Normally, a twisted wire pair and a balanced input would have rejected the crosstalk, however the 70 Volt amplifiers being used had unbalanced vs. balanced inputs and did not support common mode rejection.

The Relationship Between Output Impedance, Input Impedance and Signal Transfer

Maximum signal transfer is important when you are driving a device that will convert the signal to another form or level, like a music source connecting to an amplifier. The signal provided does not perform any actual work, it simply sources the device that <u>will</u> perform work.

This section will show you that efficient signal transfer requires that the source output impedance be negligible relative to the impedance of the input that is being sourced. *For efficient signal transfer, the source should "look up" to the load.*
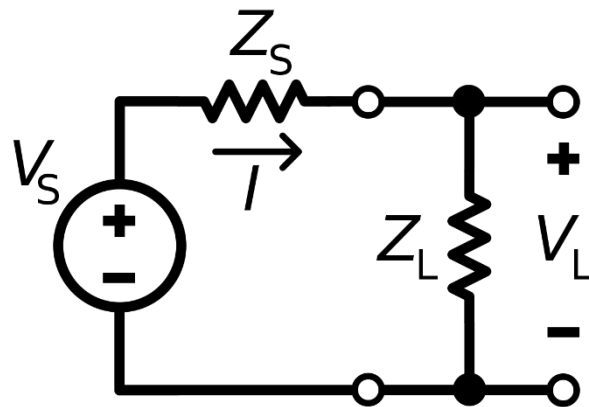
V_s = the audio source

Z_s = the audio source output impedance

Z_l = the input impedance of the load (amplifier, speaker, etc.)

The amount of source signal voltage, audio in this case, that occurs across Z_l, the input impedance of the load, is determined by:

$$Vl = Vs \left(\frac{Zl}{Zl + Zs} \right)$$

Example 1: When the source impedance (Z_s) is *less* than the load impedance (Z_l) then *most* of the source signal voltage will occur across the load:

If $V_s = 10_{vac}$ and $Z_s = 8\Omega$ and $Z_l = 45\Omega$ then $V_l = 8.49_{vac}$

$$Vl = 10vac \left(\frac{45\Omega}{45\Omega + 8\Omega}\right) = 8.49vac$$

Example 2: When the source impedance (Z_s) is *higher* than the load impedance (Z_l) then most of the source signal voltage will be lost across the source output impedance:

If $V_s = 10_{vac}$ and $Z_s = 8\Omega$ and $Z_l = 2\Omega$ then $V_l = 2.000_{vac}$

$$Vl = 10vac \left(\frac{2\Omega}{8\Omega + 2\Omega}\right) = 2.000vac$$

Example 3: When the source impedance (Z_s) is *equal* to the load impedance (Z_l) then the source signal voltage will divide equally and ½ of the source signal voltage will occur across the load.

If $V_s = 10_{vac}$ and $Z_s = 8\Omega$ and $Z_l = 8\Omega$ then $V_l = 5_{vac}$

$$Vl = 10vac \left(\frac{8\Omega}{8\Omega + 8\Omega}\right) = 5vac$$

Maximum *signal* transfer occurs when the audio source impedance = zero. However, that is not possible, so for maximum signal transfer we try to use sources with very low output impedance relative to the load input impedance (Z_l).

Note that is some cases maximum signal transfer is not ideal, for example, signal attenuation will be required if the source provides an excessive signal with respect to the load.

The Relationship Between Output Impedance, Input Impedance and Power Transfer

Maximum power transfer is important when a source is directly driving a load, like an amplifier driving a speaker. The load <u>depends</u> on the source for its ability to perform work (i.e. moving the air to create sound waves).

In realistic situations, meaning the source does not have zero output impedance, **Maximum Power Transfer** occurs when the source and load impedances are the same. Using the last equation shown in the last section, let's evaluate impedances above and below 8 ohms.

First, when the load (Z_l) matches the source (Z_S)

If $V_s = 10_{vac}$ and $Z_s = 8\Omega$ and $Z_l = 8\Omega$ then $V_l = 5_{vac}$

$$Vl = 10vac\left(\frac{8\Omega}{8\Omega + 8\Omega}\right) = 5vac$$

The power delivered to the load (Z_l) =

$$\frac{5vac^2}{8\Omega} = 3.125\ Watts$$

Next, when the load (Z_l) is greater than the source (Z_S)

If $V_s = 10_{vac}$ and $Z_s = 8\Omega$ and $Z_l = 12\Omega$ then $V_l = 6_{vac}$

$$Vl = 10vac\left(\frac{12\Omega}{8\Omega + 12\Omega}\right) = 6vac$$

The power delivered to the load (Z_l) =

$$\frac{6vac^2}{12\Omega} = 3.000 \ Watts$$

3.000 Watts is less than 3.125 Watts that was delivered when Zs was equal to Zl

Lastly, when the load (Z_l) is less than the source (Z_S)

If V_s = 10$_{vac}$ and Z_s = 8Ω and Z_l = 4Ω then V_l = 3.33$_{vac}$

$$Vl = 10vac \ \left(\frac{4\Omega}{8\Omega + 4\Omega}\right) = 3.33vac$$

The power delivered to the load (Z_l) =

$$\frac{3.33vac^2}{4\Omega} = 2.772 \ Watts$$

2.772 Watts is also less than 3.125 Watts that was delivered when Z_s was equal to Zl

This section exemplifies that efficient power transfer requires that the source output impedance *be equal* to the impedance of the load that it is driving.

25/70.7/100 Volt Speaker Impedance

The chart found here indicates speaker line load (in Watts) vs. speaker line impedance for various "constant voltage" amplifiers. Note that for any given voltage, doubling the power (1 Watt to 2 Watts, 5 Watts to 10 Watts) halves the impedance and, conversely, halving the power (2 Watts to 1 Watt, 10 Watts to 5 Watts) doubles the impedance.
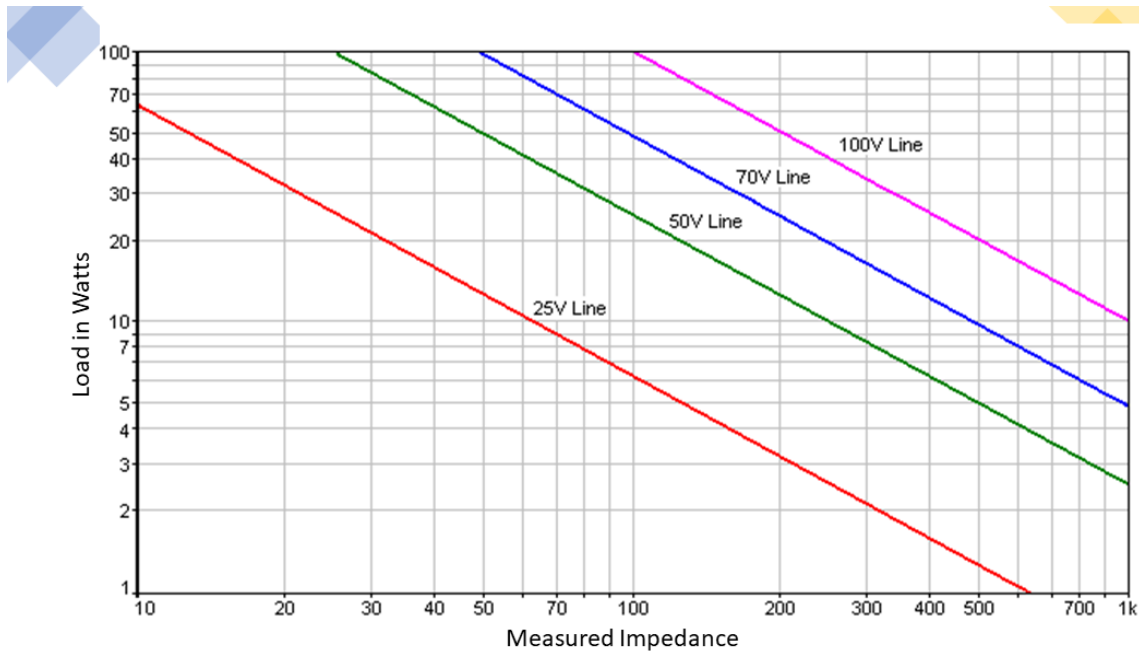
The term "constant voltage" is a misnomer because, in real practice, central amplifiers never deliver constant voltage and therefore, do not come close to delivering the level of power expected for any given speaker line load.

Impedance Meter vs. Ohm Meter

Electronic circuits respond differently to AC vs DC current.

Ohm meters use a DC current to measure a circuit's resistance. The measured resistance is a circuit's opposition to DC current.

Impedance meters use an AC current at a fixed frequency to measure a circuit's total impedance.

Since speaker circuits are sourced with audio signal, which is an AC signal, the true measure of the load presented to the audio source is the circuit's *impedance*, therefore, it should be measured with an *impedance* meter.

What is Impedance?

All speaker circuits have some level of inductance, capacitance and resistance.

Inductance responds to AC current as a function of the AC signal's frequency (Hertz). The "resistance" of a circuit's inductance **increases** as the AC current's frequency increases. This dynamic "resistance" is called reactance.

$$X_L = 2\Pi f L$$

Where f = frequency in Hertz, π ≈ 3.14159 and L = inductance in henries.

Capacitance also responds to AC current as a function of the AC signal's frequency (Hertz). The "resistance" of a circuit's capacitance **decreases** as the AC current's frequency increases. This dynamic "resistance" is also called reactance.

$$X_C = \frac{1}{2\Pi f C}$$

Where f = frequency in Hertz, π ≈ 3.14159 and C = capacitance in farads.

Pure resistance responds to AC and DC current independently of frequency.

When an AC current is applied to a circuit, the sum of the circuit's resistance and reactance is referred to as **Impedance**. The wattage load is then determined by dividing the applied audio signal voltage^2 by the load impedance ($P=E^2$/Impedance).

Voltage Drop/Signal Loss

Whenever electrical current, including audio signal, flows through wire, there is some loss. The loss is described by [Ohm's Law](#) which describes the relationship between voltage, current and resistance.

Voltage, or signal. is provided by a source. It may be an amplifier, a low-level audio output, a power supply, a generator, batteries, solar panels, etc.

Resistance is provided by the output characteristics of the source, wire, termination points, and the load (whatever is being powered – or "driven")

Current is the result of the load consuming power from the source.

In an ideal case, the source output characteristics, termination points and wire will allow current to flow unhindered (without any loss). For this to happen, all would have to have a resistance value of zero. This is never the case. Design can, however, minimize or account for the inherent loss.

All sources have some output impedance (resistance) and all wire has characteristic resistance/per given distance and all termination points have some connection resistance.

You will never realize the ideal case, however, by choosing sources with low output impedance, using manufacturer guidelines for wire size and distance and taking care to make secure connections at termination points, you will allow most of the required current to reach the load unhindered. This results in the most efficient installation possible.

The author recalls a site where, despite adhering to recommended wiring size and distance, a 24Vdc load was not operating properly. Upon measuring the voltage at the load, it was noted the 24Vdc input was only receiving 5Vdc when the load was active*. Under the same condition, the 24Vdc power supply powering the load was measuring 24Vdc. Upon evaluating all the connections, it was discovered the voltage measured across a certain termination point on a punch down block was 19Vdc when the load was active. The splice was inferior and exhibiting a high value of resistance. Once properly re-terminated, the 24Vdc load worked perfectly.

*Even idle loads may consume a small amount of current – perhaps enough to exhibit a loss condition. However, these tests are typically conducted under full load conditions.

Speaker Line Voltage Drop and Power Loss

The Formulas

Whenever 2 or more components of an electrical circuit are connected by wire, and current moves between the components, there will be loss. Ohm's Law, a fundamental law of electronics, states that Voltage (E) is the product of Current (I) and Resistance (R). This is commonly written as E= I x R.

Power is the product of Voltage (E) and Current (I). This is commonly written as P = I x E.

Using simple algebra, we can move the variables of these equations to create:

$E = I \times R$ - - - - → $P / I = I \times R$ - - - - → $P = I^2 \times R$

$E / I = R$

$E / R = I$ ⟶ $P / E = E / R$ ⟶ $P = E^2 / R$

$P = I \times E$

$P / I = E$

$P / E = I$

These formulas are used in determining the amount of voltage lost in speaker wiring.

Loss is unitless

Loss is independent of the source or amplifier's output voltage and is a function of the wire resistance and speaker impedance(s).

In simpler terms, if you have an 8-ohm speaker connected 600 feet from an audio source and you calculate that the speaker will lose 4-dB of audio due to the resistance of the 600 feet of wire, it will lose that 4-dB of audio regardless of the audio source's output level.

Applying the formulas

Audio sources, like amplifiers, provide signal voltage (E), to a speaker circuit.

The signal moves through the resistance of the wire and the speakers (R) as a current (I). Both the wire and the speakers dissipate Power (P).

The power dissipated across the wire generates undesired heat. *This means that less power is available to the speakers.*

The power dissipated across the speakers generates the desired sound.

Speaker line loss is dependent upon:

        a) the power required by the speaker
        b) the resistance of the wire
        c) the wire distance between speakers.

Once the amount of voltage across a speaker is determined by use of circuit analysis utilizing Ohm's Law, the power formula $P = E^2 / R$ is used to determine the actual power available to the speaker. Loss may be described as a percentage loss or in terms of dB. There's not a lot of detail here, but as you'll see, you don't really have to do the math.

Resistance of a speaker?

Speakers coils have frequency dependent resistance or "impedance"

$$XL = 2\pi fL \qquad f \text{ is the frequency } L \text{ is inductor size in Henrys}$$

Since the impedance of speakers, or their matching transformers, varies by the frequency of the audio presented to them, a frequency in the middle of the human hearing range is used to determine speaker impedance in order to simplify calculations. 1KHz is a common selected value for these calculations.

Calculating loss for speaker lines is simply a matter of calculating how much loss will occur in each "node" and subtracting all the losses in nodes between the source (amplifier) and any given speaker to determine the actual signal voltage that the speaker will receive. Then, the power formula $P = E^2 / R$ is used to determine the actual power available to the speaker.

dB loss is determined by:

$$10 \; x \; \log \frac{Actual \; Speaker \; Power}{Speaker \; power \; with \; no \; signal \; loss}$$

The more current (I) that flows through a resistance (R), the more voltage (E) loss will be measured across the resistance. In other words, increasing the speaker load requires more current and results in more loss. If the excessive loss is an issue, you simply use a heavier gauge wire to reduce the wire resistance.

In centrally amplified systems (25 or 70.7 volt, etc.) once the system audio has been balanced, either by way of the charts (see below) or actual calculations, it's not uncommon for speakers to be changed added or removed. Since the audio level at each speaker is dependent upon total speaker line loss, this changes the dynamic of the whole system and can force a requirement to have the system rebalanced.

By design, self-amplified or VoIP speaker systems do not experience this issue.

Sound complicated? It's not. It's just repetitive math. And what's better at repetitive math than a computer? Check out the SoundDesign Program or its instructional videos.

Most speaker manufacturers publish "rule of thumb" guidelines for AWG size (American Wire Gauge) vs. number of speakers vs. wattage rating or type of speakers vs. maximum recommended distance.

They determine the values of these charts by making educated assumptions as to the wire distance between speakers, ambient temperature (which has an effect of wire resistance), and an acceptable amount of loss.

Here is the wire gauge guide provided by Valcom, Inc. for its self-amplified speakers. For these systems, audio and power are provided over inexpensive UTP (2-pairs) and the distance from the power supply, shown in the chart below, is the only distance with which you need concern yourself.

Audio on 24 AWG UTP can travel for miles in most cases. This small gauge wire is easy and inexpensive to use for long and short distances

*VPU = Valcom Power Unit

Power Pair Wire Run								
Number of Speakers/Horns Per Power Run					Power Run Wire Length in Feet (meters)			
1 VPU* Speakers	4 VPU* Speakers	Flex Horns	5-Watt Horns	15/30-Watt Horns	24 AWG	22 AWG	20 AWG	18 AWG
4	1	-	-	-	1000' (304 m)	1600' (487 m)	2500' (762 m)	4000' (1219 m)
7	2	1	1	-	500' (152 m)	800' (243 m)	1280' (390 m)	2025' (617 m)
15	4	2	2	-	250' (76 m)	400' (122 m)	640' (195 m)	1010' (308 m)
30	8	4	4	1	125' (38 m)	200' (61 m)	320' (98 m)	500' (152 m)

Here is a loss chart provided Clarity by Valcom.

Maximum Wire Run Length (Feet) For 20 % Power Loss   REV 2/95

	Speaker Load	dBm	Wire Size (AWG)								
			# 24	# 22	# 20	# 18	# 16	# 14	# 12	# 10	# 8
70.7 Volt	10 Watt	+ 40	1149	1827	2906	4620	7345	11683	18576	29532	46959
	20 Watt	+ 43	574	913	1453	2310	3672	5841	9288	14766	23480
	40 Watt	+ 46	287	456	726	1155	1836	2920	4644	7383	11740
	60 Watt	+ 47.8	191	304	484	770	1224	1947	3096	4922	7826
	80 Watt	+ 49	143	228	363	577	918	1460	2322	3691	5870
	100 Watt	+ 50	114	182	290	462	734	1168	1857	2953	4696
	200 Watt	+ 53	57	91	145	231	367	584	928	1476	2347
25.5 Volt	10 Watt	+ 40	149	237	378	601	955	1519	2416	3842	6109
	20 Watt	+ 43	74	118	189	300	477	759	1208	1921	3054
	40 Watt	+ 46	37	59	94	150	238	380	604	960	1527
	60 Watt	+ 47.8	24	39	63	100	159	253	403	640	1018
	80 Watt	+ 49	18	29	47	75	119	190	302	480	764
	100 Watt	+ 50	14	23	37	60	95	152	241	384	611
	200 Watt	+ 53	7	11	18	30	47	76	120	192	305
	4 Ω		9	14	23	37	58	93	148	236	376
	8 Ω		18	29	46	74	117	187	297	473	752

The 20% loss chart above assumes that the entire speaker load is at the end of the wire run.

Copper Wire Resistance Chart

AWG	Diam. (mils)	Circular mils	Ohms/1000ft
0000	460	212000	0.050
000	410	168000	0.063
00	365	133000	0.077
0	324.85	105531	0.096
1	289.3	83694	0.1264
2	257.6	66358	0.1593
3	229.4	52624	0.2009
4	204.3	41738	0.2533
5	181.9	33088	0.3915
6	162	26244	0.4028
7	144.3	20822	0.5080
8	128.5	16512	0.6405
9	114.4	13087	0.8077
10	101.9	10384	1.018
11	90.7	8226	1.284
12	80.8	6529	1.619
13	72.0	5184	2.042
14	64.1	4109	2.575
15	57.1	3260	3.247
16	50.8	2581	4.094
17	45.3	2052	5.163
18	40.3	1624	6.510
19	35.9	1289	8.210
20	32.0	1024	10.35
21	28.5	812	13.05
22	25.3	640	16.46
23	22.6	511	20.76
24	20.1	404	26.17
25	17.9	320	33.0
26	15.9	253	41.62
27	14.2	202	52.48
28	12.6	159	66.17
29	11.3	128	83.44
30	10.0	100	105.2

Voltage Surge and Lightning Protection

Depending on where you live, providing [voltage surge and lightning protection](#) with the systems you install may, or may not, be routine. If your installation uses aerial or buried cable to connect buildings, then, as a minimum, those circuits should feature protection on both ends.

Plan on protecting the rest of your Valcom system to the extent that you routinely protect other low voltage systems.

There are many brands of surge (over-current/over-voltage) protection devices and we do not endorse one over the other as most Valcom systems never suffer from this type of damage.

When choosing surge protection, be certain to choose products with the correct protection value for your application. For protecting Valcom analog talkback and call button circuits, or Valcom 24VDC power supply outputs 30V protection is a good choice. Refer to the link below:

http://www.diteksurgeprotection.com/products/222-dtk-s30b.html

For Valcom one-way line level audio pairs a lower protection voltage will provide better protection:

http://www.diteksurgeprotection.com/products/219-dtk-s14a.html

There are surge protectors designed for every common use, telephone tip/ring circuits, AC power, PoE network ports, etc.

In any case. The surge protector should be located as close as possible to the device that is being protected and must be directly connected to a good quality Earth Ground using a short adequately sized braided or stranded wire, or as recommended by the manufacturer. The idea is to minimize the cable resistance to ground and provide as much wire surface area as possible to overcome the effects of [skin effect](#)*.

Improperly selected or installed surge protection devices will yield less than desirable, or no protection at all.

> *High frequency surges will tend to travel along the surface area of a conductor. Shielded or braided cable provide more surface area. Shielded or braided cables represent a lower impedance to the high frequency surge vs. an equivalent sized solid cable.

Troubleshooting – General Approach

This document is not specific to one product and its content will not apply to all troubleshoots. The document is intended to provide a list of best practices designed to help avoid troubleshooting in the first place. It is also intended is to provide a summary of tools needed for general troubleshooting and then to provide techniques for applying those tools to a variety of trouble scenarios. There are hyperlinks throughout the PDF version that will lead you to additional resources.

Witness the trouble yourself

Witness reported troubles firsthand. It is quite common for trouble reports to be inaccurate or incomplete. For example, you may receive a report that users are unable to make announcements in a certain building, where in reality; they are dialing an incorrect access code that will not allow them to make announcements anywhere.

If you can't duplicate the reported trouble, perhaps it was a false report. The author recalls a trouble where a facility's neighbors reported an all clear announcement one morning at 3 a.m. Assuming it was from the Valcom system, the technician checked the system logs and found no evidence of the system all clear being activated. As it turns out, the police department had been searching for a felon in the area and announced an "all clear" over their loudspeaker that morning. Assume Nothing.

Besides, if you can't duplicate the trouble to witness it, how are you going to know if you've fixed the trouble?

Divide and Conquer

As is true of most top tier facility systems, Valcom communication systems are comprised of multiple subsystems. Trying to troubleshoot an issue from the perspective of the entire system can be overwhelming. Subsystems are typically much easier to assess. Often, a logical approach to troubleshooting

involves isolating and testing these subsystems individually to determine the cause or causes of the trouble.

Initially attempt to troubleshoot by changing one variable in the subsystems at a time. For example, if a speaker does not work properly and the trouble is resolved by moving the speaker to a different speaker output and a different cable in one step, you won't know if the trouble was caused by the cabling or the initial speaker output.

Occasionally, however, there <u>are</u> multiple contributing factors and subsystems <u>must</u> be evaluated in various combinations. The subsystems may include the actual Valcom equipment, the telephone system, the network, the cabling, amplifiers, etc.

This document will show you techniques for evaluating many types of subsystems:

a) Given the right tools, the integrity of speakers and speaker cabling is easily evaluated while disconnected from the system.
b) Clock or common speaker circuits are easily reduced to the bare minimum, one clock or one speaker, to check for cabling issues. For example, if undesired noise is heard from common speakers, temporarily connecting just one of those speakers next to the audio source with a short cable can prove if the noise is being induced in the field wiring.
c) System audio outputs may be affected by connected subsystems (speaker lines, amplifiers, etc.). Assessing these audio outputs with and without the subsystems is very simple.
d) System relay outputs can be tested for proper operation by removing the load and checking with a meter.
e) Functions associated with date and/or time may be forced for testing by changing a system's date and time as required.
f) Power supplies should produce adequate, quiet voltage at full load. It's easy to check the supply output under full load (perhaps an all call announcement or a bell tone broadcast everywhere) and to verify that the output is stable and quiet.

Remember that while some troubles are caused by malfunctioning equipment, more often, the problem is due to environmental, design, or installation issues. Assume nothing.

Intermittent issues have been the result of cleaning services temporarily unplugging equipment to plug in their vacuums, once they are done the trouble magically clears. For IP based equipment, an IP address conflict with a PC that is switched on and off has been known to cause intermittent issues. The trouble would only occur when the PC was turned on.

Crosstalk issues have been the result of installers adding 25-volt transformers when none were required.

Noise issues have been the result of equipment being installed next to a facility's main ac power transformer.

We've even experienced troubles caused by newly purchased defective CAT 6 UTP! Everything is a variable. Assume Nothing!

Analyze the troubles and patterns may emerge to lead you to their cause. What changed when the troubles began? Network maintenance? Power outage? Something new installed? Are the reported troubles occurring:

a) Within specific buildings or areas?

b) From specific wiring closets?

c) With specific equipment?

d) With specific people?

e) On specific cables?

f) With work done by a specific technician?

g) On certain days?

h) At certain times of the day?

i) After certain events?

j) Before certain events?

All the Usual Suspects

Solutions to troubles are often found by a simple examination of the equipment involved, or the circumstances around which the trouble occurs:

a) Are the system power supplies plugged into working ac receptacles?

    a. Is there a switch or circuit breaker that may be turned off?

b) Is the polarity of the power supply connections correct?

c) Are system amplifiers turned up?

d) Are other system volume controls turned up?

e) Are all wire connections and connectors tight?

f) Are system network ports connected to the designated network switchports? Are those switchports part of the correct VLAN?

g) Is all the equipment accounted for?

h) Were there recent changes to the network or phone system being used to support the system?

i) Is the user dialing the correct codes and using the correct phone?

j) Is the trouble related to a new user that is unfamiliar with system operation?

k) Is the user holding the telephone/microphone correctly? We've seen many reports of low audio cause by users hold the originating instrument too far from their mouth.

Remove Variables

For troubles that affect a whole system, or an interrelated portion of a system, a good method of tracking down a trouble is to remove variables and test to see if the trouble clears. For example, let's assume that you have an intercom comprised of a main processor card, 3 speaker cards, and a relay card.

In this system, there is a background noise of some sort on all of the speakers when the system is idle.

To troubleshoot, remove unnecessary variables starting with least significant.

Since the noise is present when the system is idle, remove any system audio inputs, one at a time, and evaluate the result:

Microphones[1] and their cables

Telephone and telephone system connections

Music sources[1] and their cables

Next remove any unnecessary components and evaluate the result:

In this case, the relay card and 2 of the speaker cards can be removed.

Finally, eliminate necessary, but easily substituted components

Remove all speaker cabling and connect one speaker on a short piece of wire right in the equipment room.

Replace ac receptacles with a fully charged UPS.

Now you have reduced the variables to the main processor card, one speaker card and one speaker. By using the UPS for power, you have even removed the variable of noise on the ac voltage!

If the trouble persists, remove the environmental variable by physically moving the remaining equipment to a different location. Environmental variables, to name a few, includes the ambient temperature of the mounting location, noisy "dirty" ac voltage, EMF from surrounding equipment and radio frequency interference.

If the trouble ceases when the equipment is initially powered, but fails after time, it may be an internal thermal issue.

Using this method will usually put you on the path to discovering the root of a trouble. But be aware, occasionally, troubles are caused by more than one variable and it becomes necessary to eliminate and replace variables in combinations. For example, let us assume that removing two of the speaker cards resolves the trouble, but replacing either one creates the trouble. However, when the system is connected to the UPS, the trouble is gone regardless of which cards are installed. This could be caused by 2 ground loops, one on the cabling of each suspect speaker card.

Keep good notes of the combinations you have tested and the results. In the heat of troubleshooting, it is quite easy to lose track of what you've already done.

[1]In the course of troubleshooting, especially noise and EMF (RFI) issues, it's good practice to short all unused audio inputs (music input, microphone inputs) with a very short piece of wire. Before doing so, verify that the inputs are not connected to any other circuit and have no measurable voltage across their connection points.

Example of removing variables:

The author visited a site once where dual input self-amplified speaker were used. One of the inputs was connected to the public-address equipment and one was connected to a room projector to reinforce its sound.

The cable for these speakers was loosely laid on top of the ceiling grid. The facility had a welding shop and every time they used the TIG welder, there was a cracking noise from all the speakers.

The initial assumption was EMF from the TIG Welder was causing the trouble (Electric welders are notorious for creating EMF)

Troubleshooting consisted of documenting all combinations of speaker inputs (shorted, disconnected, connected) and power. Two 9-volt batteries were used to eliminate the dc power supply. It was quickly discovered that any combination of power, PA system input and projector audio input that included the projector audio resulted in the trouble. The projector was either affected by the TIG welder's EMF or its ac grounding was the suspect.

Even programming can be a variable. It's not common, but occasionally combinations of system programming can cause ill effects. To determine if this is the case, make a backup of the system's programming and return it to factory defaults. Add the minimum programming necessary to prove or disprove that the programming is causing the trouble.

Step Away

Sometimes the key to resolving a trouble is to step away for a bit. It's not uncommon to miss the obvious when you are engrossed in a specific aspect of a trouble. Remove yourself from the trouble and clear your mind. This will often lead to a resolution. The author recalls a site where a technician was attempting to modify the rate of text-to-speech audio. She adjusted and tested after each with no results. During her lunch break she realized that she was not testing with real-time text-to-speech, she was testing with a wav file she had previously created from text-to-speech.

Check for yourself

If the results of troubleshooting don't seem to make sense, and you are partially relying upon other technicians' tests or other's reports of what troubleshooting steps have and have not been completed, verify former tests and results firsthand. Assume Nothing.

Clear your thoughts

Seen this trouble before? Overconfidence can breed carelessness. Familiar troubles may not have the usual origins. Take a look at the trouble as if you've never encountered it before.

Start over

Sometimes the best way to solve a trouble is to start over. When a trouble that you assumed would be easy to correct turns into something requiring more effort, it's easy to lose track of what steps have been taken and their results. Start over and keep good notes of what has been evaluated and the results.

Know Your Scope of Work

It's not unusual for new equipment to be interfaced with previously installed, sometimes rather dated, equipment. Understanding the scope of work for the newly installed equipment and where your responsibility begins and ends is very important.

We've seen technicians spend quite a bit of time trying to deduce why the new equipment they've just carefully installed is not yielding the expected results, when in fact the new equipment is doing exactly as it should and the trouble is related to the existing equipment with which they've had to interface.

There is always a reason why end users contract new equipment upgrades or replacement, it is typically because the old equipment is no longer providing satisfactory results. However, by reutilizing parts of the old system, they potentially introduce troubles to the new system.

1) The accuracy of failing electromechanical clocks with worn gears will not increase because you've replaced the master clock.
2) Intermittent push buttons will still be intermittent when connected to new equipment.

3) Sticking contacts from monitored equipment will still stick when connected to new equipment.
4) Speakers with broken paper cones will not sound better when connected to new amplifiers.

This is where understanding the scope of your work, combined with the troubleshooting techniques you'll find in this document come into play.

The author recalls a site where relays from Valcom equipment where being utilized to operate 40+ year old electromechanical door locks. These same locks provided a contact closure when the door was unlocked or ajar. These contacts were connected to Valcom equipment to provide indication that the door was unsecured.

Several of the doors would not unlock as designed. However, monitoring the relay outputs of the Valcom equipment verified that the activation contact closures were indeed operating as designed. Therefore, the locks were at fault. The scope of work was to provide activation for the lock.

On this same site, several doors would intermittently indicate that they were unsecured when they were actually closed and locked. By monitoring the inputs of the Valcom equipment where the "door ajar" contacts were connected, it was obvious that the "door ajar" contacts were sticking closed from time to time, thus causing the fault.

Maintaining an Undocumented Site

End user and integrator personnel often change. On occasion, you may be asked to maintain unfamiliar equipment or to determine its capabilities. You may occasionally get lucky on a troubleshoot, but you cannot successfully maintain what you do not understand. To facilitate this:

1) Perform a site survey to identify and record on site equipment labeling and MAC addresses (if applicable), part numbers, locations and quantities.
2) Record the locations of any labeled cabling and terminations
3) Determine and record cable routing to the best of your ability
4) If labeling is absent, record and label as you identify equipment, cabling and terminations
5) Retrieve or obtain current backups of system programming. If executed properly, the programming will include descriptions of equipment and I/Os
6) Archive current backups of system programming by site name, date and time. Never overwrite these.
7) Interview any recent users to obtain their understanding of intended system operation
8) Contact Valcom to determine if the original installer/designer shared or asked us to archive system details
9) Keep thorough notes as you analyze the system

Clearly document your findings to avoid future maintenance delays.

Refer to the Documenting the Site section of this guide.

Test tools and resources you will need

Current Product Manuals

When manufacturers wish to advise customers of product tips, proper installation and caveats, they quite often give notice in the product's manual. The author of this document has observed many long troubleshoots that could have been avoided by simply referring to the current documentation.

Valcom Engineered Solutions Documentation – http://www.ValcomES.com
Valcom General Line Documentation – http://www.Valcom.com

Lineman's handset

In communication systems accessed via FXO ports or Loop Start C.O. Line/Trunk ports, a lineman's handset may be used in place of the telephone system to determine if it is the source of volume, noise or access issues. Simply connect the lineman's handset directly to the unterminated FXS or Tip/Ring I/O of the Valcom system (the I/O that provides dial tone), go off-hook with the lineman's handset and dial the access digits required by the Valcom unit under test (VUUT). If the test is successful then the VUUT is operating properly and cabling to the telephone system, or the telephone system itself, should be investigated as the source of the trouble.

Always test phone system inputs and outputs with both cabling attached and cabling removed to identify cabling related faults. Direct short circuits will kill a signal, high impedance in cabling will reduce the level of a signal and open circuits will prevent the signal from passing.

System ports that provide dial tone **must not** be directly connected to any other system ports that simultaneously provide dial tone.

Audio Monitoring

Need to hear low level audio signals in a cable or from an audio source?

A lineman's test in monitor (hi-z) mode may be used to:

a) "listen" to dc power supply outputs to verify that they are quiet
b) verify the presence of audio on cabling to Valcom Speakers
c) verify the presence of audio from any system music sources.

Remember to test at the point of entry (on the actual speaker or music input) and/or point of exit (on the music source or VUUT audio output).

To identify cabling short or open circuits, always test for audio on the terminated audio source point (to check field connections for short circuits) and on the unterminated destination cabling (to check wiring for open circuits). Direct short circuits in cabling will kill a signal, high impedance in cabling will reduce the level of a signal in the receiving device, and open circuits in cabling will prevent the signal from passing to the receiving device. Refer to the section on impedance meters for instructions on measuring cable impedance.
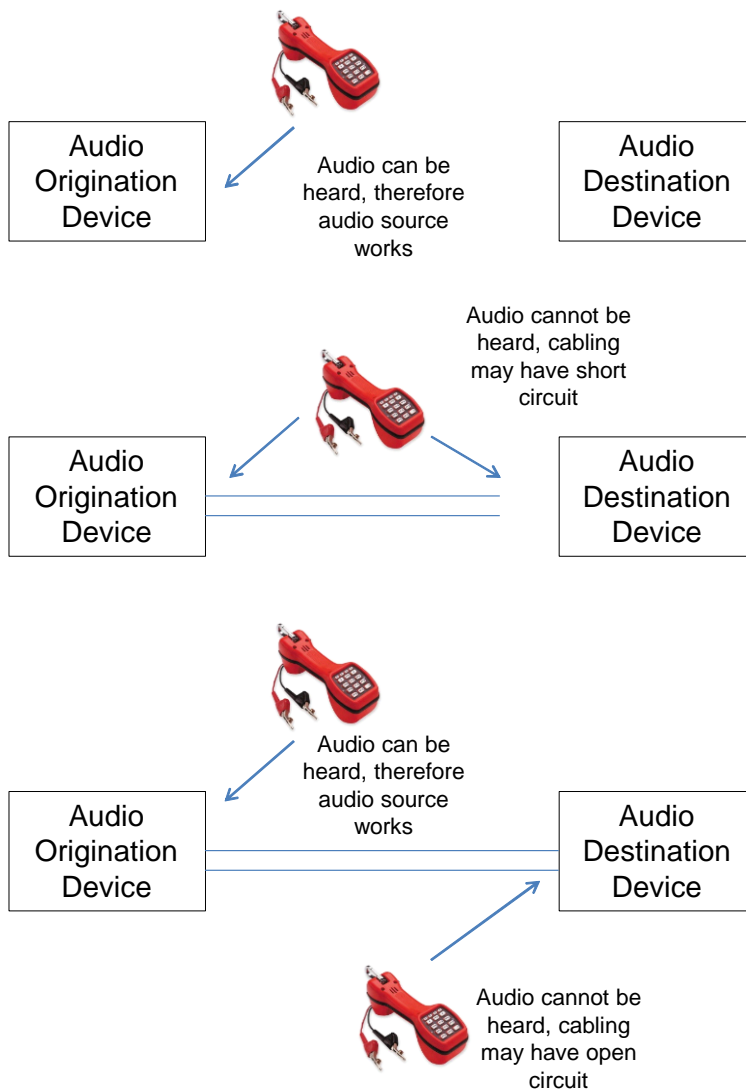
Always check your lineman's handset on a known working circuit to determine if it is functioning properly. The author of this document has encountered more than one defective lineman's handset.

Toner

Cable toners are marketed for identifying cables. They generate an audio tone in the cable and an inductive amplifier is then used to identify the cable at some other point. They are also useful in the role of portable audio source to check audio inputs such as music inputs, self-amplified speaker runs and line level amplifier inputs. Some integrators routinely test, and document speaker audio runs as they are installed by powering the amplifiers or self-amplified speakers and sending tone from the beginning of the audio wire pair. Great idea!

Volt Ohm Meter

Volt Ohm Meters are commonly used to check for voltage from power supplies and to check for dc short circuits in cabling. Voltage measurements should always be made under maximum system load conditions. In public address systems, a solid mid frequency tone playing everywhere at a volume mimicking the maximum required audio level will create a maximum load condition. True RMS Volt Ohm meters are a better choice for measuring ac voltage. Voltage level measured at different points on a cable will drop as distance from the power source increases. This is due to the relationship described by Ohm's Law. If you are unfamiliar with using a Volt Ohm Meter (aka Multimeter), there are a myriad of explanatory videos on the Internet.

System outputs that provide voltage **must not** be connected in parallel to any other outputs that simultaneously provide voltage.

PoE Load Meter

PoE (Power over Ethernet) testers are commercially available to evaluate the load on PoE switch ports.

Impedance Meter (Goldline)

Audio is an ac signal. Speakers and cabling have capacitive and inductive characteristics which react to ac signals by changing their opposition to current flow or impedance. Therefore, properly checking a subsystem's (speaker line, audio input, etc.) response to audio must be made with an ac signal. Impedance meters measure the ac impedance of cable pairs, loudspeakers and entire distributed loudspeaker systems. This allows you to determine the actual wattage of speaker lines or to find ac short circuits. Some will also calculate the wattage that will be produced on an installed loudspeaker system.

Volt Ohm meters *are not* a substitute for impedance meters as the capacitive and inductive characteristics of speakers and cabling do not react to the meter's dc voltage output.

To measure cable impedance, simply disconnect both the source and load from the cable. Leave connectivity parts (blocks, jacks) in place. Short the point where the load would normally connect and measure the impedance through the loop from where the source would normally connect.

Non-Configured Multiport PoE or PoE+ switch

Much like a lineman's handset can be used to prove or disprove the telephone system's potential role in troubles, a non-configured multiport PoE switch (test switch) can be used to prove or disprove a network's potential role in troubles. If a piece of Valcom equipment is not working properly through the network, then connecting it through a test switch removes all of the potential network variables in order to indicate if the trouble is in the network configuration/hardware or in the Valcom endpoint. PoE+ may be required for some tests. Note that changes to the endpoint's, and your PC's, IP addresses may be required for testing.

As an example, a Valcom audio gateway is at default and its IP address is 192.168.6.200. You connect this gateway to the network, scan with the VIP-102B IP Solutions Setup Tool, but the gateway is not discovered. Is there an issue with the gateway, the PC or the network setup? You remove the gateway from the network, plug it into your non-configured multiport PoE switch, and then plug your PC into the non-configured multiport PoE switch. You change the IP address of your PC to 192.168.6.201, and then scan again with the VIP-102B IP Solutions Setup Tool. Voila, the gateway appears in the scan. Now you've confirmed that there are issues in the network setup, in the network ports, or the original Ethernet cables.

In cases where the originating and receiving endpoints are not dependent upon other systems and are configured with unique static IP addresses in one subnet, they may easily be moved from the facility's LAN to a non-configured multiport PoE or PoE+ switch for testing. For example, if you experience trouble making an announcement through one or more IP speakers from a telephone connected to an FXS gateway, and the speakers and the FXS gateway all have static IP addresses in the same subnet, simply move one or more of the speakers and the FXS gateway to the test switch. If the trouble clears when the equipment is connected to the test switch, the facility LAN setup will require evaluation.

Network Tap

Some network traffic analyzation requires [port mirroring](). One easy way to accomplish port mirroring is with a network tap such as a Dualcomm DCSW-1005PT 10/100 Ethernet Network TAP w/PoE Pass-Through.

2-way Radios

Testing internal communication systems often require multiple technicians working together in different parts of the facility. For example, one technician may be walking around a facility verifying that speakers in different areas receive group announcements while a second technician is at the head end to initiate the announcements. Having the ability to instantly communicate can expedite the testing process significantly.

Long Range cordless phones ([Engenius]())

Long range cordless phones offer a convenient method of testing out a Valcom communication system. They may be connected to an FXS or Tip/Ring input of the system to allow technicians to initiate announcements from any part of the facility. This can save massive amounts of labor time.

1:1 Audio Isolation Transformers

Audio Line Isolation Transformers, such as Valcom's [VMT-2](), are used to electrically isolate audio outputs from audio inputs in order to prevent ground loops.

Ground Lift Plugs

[Ground lift plugs]() are useful when troubleshooting issues such as [ground loops]() or hum. They should only be used to identify the cause of an issue, never as a permanent solution.

Batteries

If noisy power supplies are suspected of inducing noise into a system, temporarily substitute the supplies with DC batteries (2 9-volt batteries connected in series will power most 24-volt speakers and circuits). If the batteries quiet the noise, then the power supplies are the likely noise source.
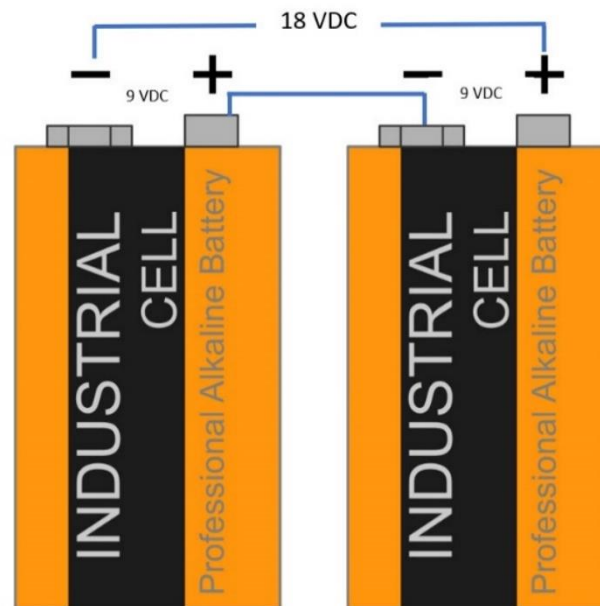
A Non-Blocked Laptop PC

You may need a laptop PC running a current version of Microsoft Windows, PuTTY and Wireshark. This is often required to program systems and assess network connectivity and setup. The PC must have an RJ45 accessible NIC and be unencumbered by restrictive "protection" software. The author has witnessed many a wasted hour caused by "protection" software preventing required software installation or blocking required ports.

Wireshark Decode Filters

Valcom has created custom Wireshark filters to decode the network analyzer's VALCFG and VAL_CC results into meaningful information. The file includes a demonstrative video.

A Hotspot

Remote tech support often requires Internet connectivity for remote sessions and firmware upgrades. A readily available hotspot can save significant troubleshooting time.

A Cellphone

A quick call to Valcom Technical Support can also save significant troubleshooting time. Please call 1-540-563-2000 from the jobsite with the resources required for troubleshooting available. Remote tech support often requires Internet connectivity for remote sessions and firmware upgrades. If your cellphone is a smartphone, then a bonus is having the ability to send pictures, video and sound recordings to help communicate troubles to technical support.

A Labeler and Permanent Marker

Labeling cables, connections and equipment as they are installed, or identified, will help in preventing connection errors and assist with future maintenance.

RJ45 and RJ11 Inline Couplers

Front   Back

Technicians sometimes find it necessary to patch in to. or temporarily extend, RJ45 or RJ11 terminated cables. Inline couplers make this an easy task.

A Sound Level Meter

A sound level meter, as the name implies, is used to measure sound levels (pressure). The sound level is displayed in terms of dBspl value. Most sound level meters have multiple weighting filters that can be applied. 'A' weighting is a standard weighting of audible frequencies and is designed to reflect the audible response of the human ear. C-frequency-weighting however is still used in the measurement of the peak value of sound pressure.

Digit Grabber

A [Digit Grabber](#) monitors and analyzes DTMF and dial pulse signaling through any telephone or wireless communications system

Your Eyes and Ears

Witness reported troubles firsthand. It is quite common for trouble reports to be inaccurate or incomplete. For example, you may receive a report that users are

unable to make announcements in a certain building, where in reality; they are dialing an incorrect access code that will not allow them to make announcements anywhere.

If you get a report that something is not working properly, and you cannot reproduce the reported trouble, have the reporting party duplicate it for you. You may see that they are simply not following the steps required for proper operation.

The author recalls a situation where an end user reported that the system time on her Class Connection ES Intercom was off by 3 minutes every morning. This forced her to correct the system time every morning. Upon investigation, it was noted that the Class Connection system was correcting to a time server every morning at 2:45 a.m., and that its time was completely accurate.

The end user would look at a battery-operated clock every morning and notice a variation between the clock's time display, which was inaccurate, and the Class Connection ES time. The only trouble was her choice of time standard! Assume nothing.

Remote Access

Negotiating remote access to jobsites via a VPN or a remote support service such as GoToAssist or Zoho Assist can help you to solve customer site programming and configuration issues quickly from anywhere with Internet connectivity.

Useful Software Utilities

There are many useful software utilities that may assist you in troubleshooting.

Packet Sender - Free utility to for sending / receiving of ...https://packetsender.com

Packet Sender is a free utility for sending / receiving network packets. Support for TCP, UDP, and SSL.
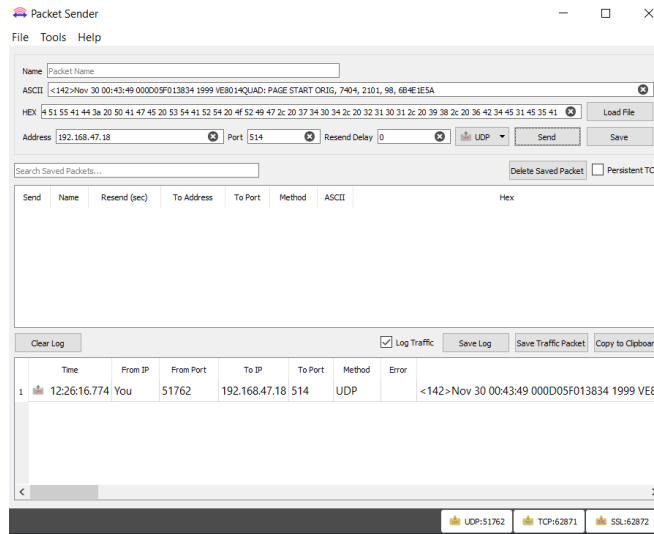
CurrPorts is network monitoring software that displays the list of all currently opened TCP/IP and UDP ports on your local computer. For each port in the list, information about the process that opened the port is also displayed, including the process name, full path of the process, version information of the process (product name, file description, etc.), the time that the process was created, and the user that created it.

Wireshark is the world's foremost network protocol analyzer. It lets you see what's happening on your network at a microscopic level. It is the de facto (and often de jure) standard across many industries and educational institutions.

PuTTY is an SSH and telnet client, developed originally by Simon Tatham for the Windows platform. PuTTY is open-source software that is available with source code and is developed and supported by a group of volunteers. You can also download PuTTY here.

Simple IP Ping Range is a small utility allows you to easily ping multiple host names and IP addresses. Simple IP/IP Range Ping simply pings each IP address and displays the number of succeed and failed pings, along with the average ping time. You can also save

entire log, open hosts, or closed hosts to text file. Program requires that Microsoft .NET Framework 4.0+ installed.

Valcom VIP Audio Tool is a program for recording and processing WAV files for use with Valcom systems.

Valcom WAV File Format Tool easily converts WAV files to the correct format for Class Connection ES systems.

MD5 is a u**ser-friendly and portable piece of software designed to help you verify downloads by comparing two MD5 checksums, as well as generate checksums.**

VIP Utility Tool This Valcom Utility should only be used under the direction of Valcom Technical Support. Improper use may render your device inoperable.

For all options, enter the IP address and password of the VIP device to be modified. Click "Use SSH" to enable SSH communication or leave unchecked to use telnet communication.

Collect Console Output will enable the Valcom device to run in debug mode with the console output directed to a text file for further analysis. The device should be rebooted when the console session is ended.

Display/Set Network Addresses will retrieve the current network address settings from the VIP device and allow changing the IP Address, Subnet Mask and Gateway. Rebooting is required for any changes to take effect.

The Ping option will send 5 ICMP Ping packets to the IP address entered in the address text box. This is useful for checking network access to a SIP server or time server. By inputting a host name, it can also be used to verify DNS name resolution.

Check Date and Time will retrieve the current date and time from the VIP device; typically used for troubleshooting clocks.

Set Date and Time will first retrieve the current date and time from the VIP device, then set the time to the value specified here. To use the time on this PC, select the Use PC Time box. Or enter a time manually using the selector. Times are set using UTC.

Download Log Files will retrieve the internal log file(s) from the VIP device, if they exist. The program will prompt for a file name on this PC to store the logs in a compressed file. The default file name is "<ipAddress>_log.zip", where "<ipAddress>" is the address of the VIP device. The name can be changed as desired.

Additionally, the utility may also be used to try to regain control of a VIP device that is rebooting frequently, preventing normal operation. The first attempt should be to simply stop the running process. If that does not break the reboot cycle long enough to remedy the

problem, then the additional step of renaming the process may be required. In both cases, additional troubleshooting steps will be necessary to fully resolve the problem.

Two methods are provided to rename a process, Auto and Specific. The default is Auto, which will attempt to determine the correct process name automatically. The Specific method requires the process file name to be entered. This would typically be used when the process is dying too quickly to be determined automatically.

If the process was renamed, the Restore Original Name option can be used to restore the file name or a firmware upload will be required to return the device to operation.

 When using the option to stop or rename the running process, this program must login to the device during the period between reboots. Therefore, it may be necessary to execute this program more than once to successfully complete the operation before the next reboot.

This utility program can edit the Engineering Debug file. This file is used to enable special functions for test and debugging. Use this option only with the assistance of Valcom Technical Support.

MultiCastDiags is a software tool for testing the multicast configuration of a network that will be hosting Valcom IP Speakers, Gateways and servers. This software simultaneously sends and receives multicast packets to/from up to 5 multicast addresses. Installing this program on 2 PCs allows verifying multicast traffic between 2 network switch ports.

The tool is very easy to use:

a) Install the latest version of Wireshark on the PCs as MultiCastDiags uses some of the same core programs as Wireshark.
b) Install the VIP-102B IP Solutions Setup Tool *and* the Valcom IP Solutions Multicast Diagnostic Tool on each PC that will be used for testing. Disable all firewalls and wireless NICs on these PCs for the duration of the testing.
c) Exit all other Valcom programs (like the VIP-102B)
d) Assign the PCs valid IP addresses for your VLAN and plug the PCs into the VLAN network ports under evaluation.
e) Set the send interval to the same value on all computers involved in the test.
f) Check "Echo Received Packets by Multicast", "Echo Received Packets by Unicast", "Receive All" and "Start" on one or more "remote computers"
g) Check "Send all" and "Receive all" and "Start" on a "control computer"
h) Allow the control computer to receive 50 or more packets from each remote computer.
i) Click "Stop", then click "Details" on the control computer to verify reception of data and see calculated average round trip times from each remote computer.

This software has been run on Windows XP, 7, 8.1 and 10, but has not been tested extensively.

Unique address/port are required for accuracy (For example, setting multiple multicast addresses to use the same port will probably not return accurate counts. The total number of packets received may be correct, but it is indeterminate which particular thread will get the delivered packet)

There is a performance penalty during startup that will cause the packet delivery time to be inaccurate. Generally, the first packet or two will show a longer RTT (Round Trip Time) due to startup delays. The average RTT becomes more realistic after running for some bit of time. Allow time for 50 or more packets to be received for accurate results.

The round-trip time also includes the time used to process the packets by the program itself, which adds some small amount to the total. If RTT is more than 40 milliseconds, which would be 20 milliseconds one-way, then VoIP audio could be affected.

Typically, one "control computer" will be set to "Send All" and "Receive All" but not to "Echo Packets Received by Multicast" or to "Include Unicast Response to Sender".

Other "remote computers" connected to different network ports under test will be set to "Receive All", "Echo Packets Received by Multicast" and to "Include Unicast Response to Sender", but not to send anywhere.

This will allow the "control computer" to test multicast traffic to and from each "remote computer" and also to calculate average RTT from each.

If unicast echoes back from a remote computer, but multicast does not, then the remote computer received multicast from the control computer but was unable to send multicast back.

If neither multicast nor unicast echo back from a remote computer, then the remote computer did not receive multicast from the control computer or was unable to send multicast or unicast back.

Refer to the following screenshots:

Valcom IP Solutions Multicast Diagnostic Tool

File   LLDP   Trace   Announcer   Help

Network Interface: 192.168.4.29
Interval (sec): 5   TTL: 64

☑ Send All
☑ Receive All
☐ Echo Received Packets by Multicast
☐ Echo Received Packets by Unicast

☐ Tag Packets with VLAN: 1
☑ Use Default Host Name
LAPTOP

Send	Receive	IP Address	Port	Sent	Rec'd	Multicast Echoes	Unicast Echoes
☑	☑	239.1.1.2	4097	0	0	0	0
☑	☑	239.1.1.3	4098	0	0	0	0
☑	☑	239.1.1.4	4099	0	0	0	0
☑	☑	239.1.1.2	4120	0	0	0	0
☑	☑	239.1.1.5	4121	0	0	0	0

Factory Defaults   Details   Start   Exit

Status:

Valcom IP Solutions Multicast Diagnostic Details

Network Interface: 192.168.96.127   Multicast TTL: 64   Start Time: 10/3/2014 2:42:
Interval (sec): 5   End Time: 10/3/2014 2:45:
Elapsed: 00:02:22

Multicast Address	Port	Sent	Rec'd	Multicast Echo Rec'd	Multicast RTT (avg ms)	Unicast Echo Rec'd	Unicast R (avg ms)		
239.1.1.4	4099	28	0	28	1.7500	28	1.857		
239.1.1.2	4097	28	0	28	1.5714	27	1.625		
239.1.1.3	4098	28	0	28	1.6429	27	1.703		
239.1.1.2	4120	28	0	28	1.7500	28	1.785		
239.1.1.5	4121	28	0	28	1.9286	27	2.037		
239.1.1.3	4098	28	0	28	0.6429	27	0.629		
239.1.1.2	4120	28	0	28	0.6786	28	1.357	192.168.96.111	WIN7PRO055
239.1.1.4	4099	28	0	28	1.3214	27	1.3704	192.168.96.111	WIN7PRO055
239.1.1.5	4121	28	0	28	1.3571	27	1.3704	192.168.96.111	WIN7PRO055
239.1.1.2	4097	28	0	28	0.5714	27	0.5926	192.168.96.111	WIN7PRO055

Export to Text File   Close

Tip – Prove that the PCs running MultiCastDiags.exe don't have any blocked ports that will skew the test results. Initially test with the two PCs connected to a simple unmanaged switch or through a cross over cable. If the multicast testing is successful   on this "bench test" but not on the actual network, then you can be certain that the network is not properly configured.

If successful paging is sporadic,

For example:

The first page is unsuccessful

The second page is also unsuccessful

The third page is successful and subsequent page attempts immediately following the successful page continue to work
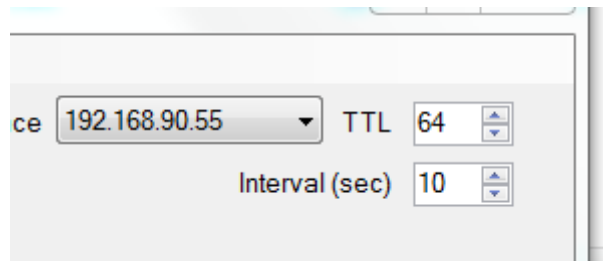
However, if you wait for some period of time, the **failure sequence** starts over again.

(or some similar pattern)

Then it is likely that the problem is that the multicast membership in the remote subnets is not being communicated up through the network to the originating subnet. Each page

opens a multicast stream (first the control, then the audio) and it all times out after some period of inactivity.

To test with Multicastdiags.exe, Start with the interval set to a value just above the time required to reinitiate the **failure sequence**, then change the interval to 5 or 10 seconds and compare the results.

If the multicast test is consistently successful at the shorter interval, but fails at the longer interval then the multicast membership in the remote subnets is not being communicated up through the network to the originating subnet.

Other MultiCastDiags Features:

Multicast Trace will query routers to find out how multicast would be handled from one point to another. This is dependent on the routers having the mtrace function built-in.

LLDP will send an LLDP packet from the PC into the switch. If LLDP is turned on in the switch, then some information from the switch may be returned. Interesting information could be things such as the port's VLAN assignment, whether Voice VLANs are in use (and which VLANS are Voice VLANs), the switch port designation, etc. LLDP uses the WinPCap library, which is installed with Wireshark. WinPCap can be installed separately, but it is easier to install Wireshark and let it do the install.

MAC/IP Announcer will send the MAC and/or IP address announcement as a unicast page to an individual device. This may be useful for locating speakers that were installed without properly documenting their location.

New in version .0900 - IGMP Information is a tool to help diagnose problems with IGMP (Internet Group Message Protocol) on a network.

A typical network will have at least one, and sometimes several, devices performing the IGMP Query function. Devices wanting to receive multicast will reply to the query with an IGMP Report. The Report will indicate which multicast addresses (and optionally, sources) the device is interested in receiving. The network switches and routers use the Report information to correctly route multicast where it needs to go.

Running the IGMP Information function requires Administrator privileges.

Valcom Paging Monitor can receive and analyze multicast audio and text packets generated by Valcom IP Paging Devices. A help menu is available in the tool.

This software tool provides diagnostics for troubleshooting purposes. This may help with situations where data capture using Wireshark is not easily done. It may also be useful to convince a network admin that there really is a problem with the network that needs to be investigated.

In addition to the Dropped Packet counter and Sequence Errors counter, there are columns for Delta and Jitter.

Delta shows the average and maximum values detected. For Valcom paging with 20 millisecond packet sizes, the ideal value for Delta is 20. Values close to that are good. If the average and max are close to 20, then the network is probably delivering the packets okay.

Jitter is a measure of how closely packets are evenly spaced. Ideally, every packet would arrive 20 ms after the previous packet. Jitter is the measure of the deviation from the ideal between successive packets. It should be close to 0 for a well-behaving network.

A video example is shown here.

[SLP-Responder](#) (Service Location Protocol)

This Valcom Utility program is intended to help diagnose and resolve problems related to Service Location Protocol (SLP) and Valcom IP Speakers. By providing an application to easily allow responses to SLP requests sent by Valcom speakers when booting up, this program may help with diagnosing SLP multicast network problems.

Service Location Protocol is designed to use multicast address 239.255.255.253 and UDP ports 427 (listener) and 437 (sender). This utility will listen for SLP requests from Valcom IP Speakers and respond. Only requests for "InformaCastConfiguration" in the "Berbee® Applications" scope are recognized and responded to.

When using this program to serve as a substitute SLP responder, the configuration will generally use this computer's network interface for the SLP IP Address field and the IP address of the InformaCast® server as the Register IP Address. The default port (8081) will usually be correct but may be changed if necessary.
When used this way, the local PC should be in the same subnet with the Valcom Speaker being diagnosed. This will provide the SLP service close to the Speaker. If the Speaker is able to register with the InformaCast® server after getting SLP information from this PC, then this would indicate a network problem with multicast.

This program may also be used to allow Valcom InformaCast® Speakers to be booted up in an environment where there are no InformaCast® servers. Typically, this can be used for situations where an InformaCast® speaker was ordered incorrectly or for troubleshooting IC speakers that have been removed from the IC environment.

For this use, both the SLP and Register IP address fields would be set to use this PC's IP address. This program will then respond to the SLP requests and provide the configuration file to the speaker. This information allows the speaker to boot up normally, after which it can be scanned and configured using the VIP-102B IP Solutions Setup tool.

A minimal DHCP server is provided for use on networks where DHCP service is not already available, or for troubleshooting connectivity from the normal DHCP service. To enable this feature, check the box to Provide DHCP Service and enter the DHCP scope parameters.

It is important to keep this program running until all speakers have been configured as necessary.

This program requires Administrator privileges to run properly. When starting the program, use the right-click menu option to "Run as Administrator". On the first run, Windows Firewall may prompt for permission to allow communication through the firewall. This must be agreed to for this program to work.

Additionally, it may be required to manually add a firewall rule to allow HTTP traffic on the port selected as the Register Port (8081 by default). This is typically done through the Control Panel -> Windows Firewall applet. Under "Inbound Rules", use the New Rule wizard to allow TCP connections on the port being used.

NetTime is a Simple Network Time Protocol (SNTP) client for Windows 95, 98, Me, NT, 2000, XP, Vista, 7, 8, 10 and Server 2003/2008/2012/2016. (32 bit and 64-bit operating systems are both supported). There is an option on the Settings menu "Allow other computers to sync to this computer" that allows the PC running NetTime to act as an NTP server.

SoundDesign is a software utility that performs many calculations involved in PA system design. It can easily determine correct wire size for speaker wiring runs and dBspl calculations vs. speaker spacing.

NetSetMan IP Address Management Utility

Are you tired of manually changing your location-based network and system configuration on your laptop computer every day? Then NetSetMan is your solution. It will do the work for you. Switch between configuration profiles for different locations instantly!

SendDataCS is a simple program to send a string of data (usually text) to an IP address destination.
The protocol can be either UDP or TCP. The "History" button will display the text messages sent, but does not store the destination address.

This program is generally useful for testing the Text Monitor function in the VE602x APP Server.

PREPARING THE SITE FOR TROUBLESHOOTING

If it becomes necessary to evaluate system performance versus customer expectations on-site, an ideal scenario for the evaluation will include:

1) A clear description of the scope of work and customer expectations.
2) A clear description of the reported anomaly.
3) The ability to witness and consistently duplicate the reported anomaly.
4) The dedicated attention of at least one qualified technician that:
   a. Can perform required tests
   b. Knows the location of all equipment
   c. Knows the routing of all infrastructure
   d. Knows the terminations of all infrastructure
   e. Knows all equipment passwords and log in credentials
   f. Has all required tools, ladders and lifts required to make any necessary adjustments or perform any necessary tests
5) Full unrestricted access to all areas involved in the reported anomaly.
6) Full unrestricted access to all relevant equipment.
7) Cell phone service or access to a telephone with long distance service at all equipment locations.
8) Wi-Fi and/or wired Internet access with a guest password. It may be necessary to connect to the network serving the Valcom equipment and to the Internet simultaneously.
9) Permission:
   a. To take digital pictures and video
   b. To connect an unrestricted laptop, or other unrestricted PC, to the network serving the Valcom equipment

If the Valcom equipment interfaces to any other systems, then reserved access to whoever manages or administers those systems must be available. These include, but are not limited to:

1) Data Networks
2) Telephone Systems
3) Radio Systems
4) Social Media Accounts
5) Websites
6) Fire Alarm Systems
7) Door Access Control Systems

How to Test Basic Functionality

It is advisable to wear hearing protection when working in close proximity to speakers and horns.

IP speakers and Horns

Connect the speaker or horn under test (UUT) to an [unmanaged multiport PoE switch](). Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet (i.e. if the IP address of the speaker or horn is 192.168.6.203, and then change the PC IP address to 192.168.6.200).

Rescan the switch. You should now be able to receive group membership information from the UUT.

Add the UUT to an audio group, set the channel output volume to -15, and update/reset the UUT to achieve a normal status. Make certain that all conflicts are cleared.

Use the Communications/Send sample page utility of the VIP-102B to send sample audio to the UUT. The UUT should broadcast the audio.

These are basic tests. If using Session Initiation Protocol (SIP) for access, then following the basic test, the UUT should be relocated to, and properly configured to work on, a network segment that is routable to the telephone system VLAN. Properly configure the SIP Tab of the product being tested and access the UUT via a SIP telephone. Refer to the SIP (Session Initiation Protocol) Troubles section for more information.

Self-Amplified Speakers and Horns

Connect your voltmeter probes to the dc voltage input of the UUT. You should measure approximately 24 or 48 vdc (it may be a negative or positive reading depending upon your meter probe orientation). The UUT will work with either polarity. Leave the voltmeter connected to monitor voltage levels throughout the test.

Verify that that volume control of the UUT is not turned all the way down.

Disconnect all cabling from the tip/ring input of the UUT.

Verify that your [toner]() (cable toner) is working by monitoring the output with your inductive amplifier or lineman's test set.

Connect the output of your toner to the tip/ring input of the UUT.

With the toner turned on, you should hear the tone through the UUT. DC Voltage should remain relatively steady.

45 Ohm Talkback Speakers and Horns

Verify that your [toner](#) (cable toner) is working by monitoring the output with your inductive amplifier or lineman's test set.

Disconnect all field wiring from the speaker/horn.

Connect the output of your toner to the tip/ring input of the speaker/horn.

With the toner turned on, you should hear the tone through the speaker.

FXS Gateways used as an input to an IP system

Connect the FXS Gateway under test (UUT) and a known working IP speaker to an [unmanaged multiport PoE switch](#). Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT and the IP speaker. Set the PC IP address to be on the same subnet as the FXS Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

If the IP speaker is on a different subnet than the UUT, then reassign its IP address so the 2 endpoints are on the same subnet (i.e. 192.168.6.201)

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add the IP speaker to an audio group. Do not add the UUT to the audio group. On the VIP-102B's Channel Tab for the IP speaker, change the output volume to -15. Remove any entries from the UUT's auto destination field on the Channel Tab(s), and update/reset to achieve normal status of both endpoints. Make certain that all conflicts are cleared.

Connect a POTs telephone or lineman's test set to the phone input of the UUT.

Go off-hook and you should receive dial tone. Dial the audio group ==*access code*== and you should be able to make an announcement through the IP speaker. Dial the IP speaker's channel ==*dial code*== and you should be able to make an announcement through the IP speaker.

Retest for each available phone input of the UUT.

Use the Communications/Paging Diagnostics page utility of the VIP-102B to verify that page start and page stop commands are being sent when the group is dialed and when the group page is concluded.

FXS Gateways used as an output to POTs telephones or equivalent

Connect the FXS Gateway under test (UUT) to an [unmanaged multiport PoE switch](#). Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet as the FXS Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add each channel of the UUT to an audio group. On the VIP-102B's Channel Tab(s) remove any entries from the UUT's auto destination field(s), and update/reset to achieve normal status. Make certain that all conflicts are cleared.

Connect an on-hook POTs telephone or lineman's test set to the phone output of the UUT.

Use the Communications/Send sample page utility of the VIP-102B to send sample audio to the group that includes the UUT. The on-hook POTs telephone or lineman's test should ring. Go off-hook to hear the audio.

Retest for all phone outputs of the UUT.

These are basic tests. If using Session Initiation Protocol (SIP) for access, then following the basic test, the UUT should be relocated to, and properly configured to work on, a network segment that is routable to the telephone system VLAN. Properly configure the SIP Tab(s) of the product being tested and access the UUT via a SIP telephone. Refer to the SIP (Session Initiation Protocol) Troubles section for more information.

FXO Gateways used as an input to an IP system

Connect the FXO Gateway under test (UUT) and a known working IP speaker to an [unmanaged multiport PoE switch](#). Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT and the IP speaker. Set the PC IP address to be on the same subnet as the FXO Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

If the IP speaker is on a different subnet than the UUT, then reassign its IP address so the 2 endpoints are on the same subnet (i.e. 192.168.6.201)

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add the IP speaker to an audio group. Do not add the UUT channel(s) to the audio group. On the VIP-102B's Channel Tab for the IP speaker, change the output volume to -15. Remove any entries from the UUT's auto destination field(s) on the Channel Tab(s), and update/reset to achieve normal status of both endpoints. Make certain that all conflicts are cleared.

Connect your on-hook lineman's handset to an analog station port of a telephone system or equivalent. Dial the station port via the telephone system and the lineman's handset should ring.

Disconnect the lineman's handset and connect the analog station port to the line input of the UUT.

Dial the station port via the telephone system and you should receive dial tone. Dial the audio group access code and you should be able to make an announcement through the IP speaker. Once again, dial the station port via the telephone system and you should receive dial tone. Dial the IP speaker's channel dial code and you should be able to make an announcement through the IP speaker.

Use the Communications/Paging Diagnostics page utility of the VIP-102B to verify that page start, and page stop commands are being sent when the group is dialed and when the phone is disconnected.

Retest for all line inputs of the UUT.

FXO Gateways used as an output to another system's tip and ring

Connect the FXO Gateway under test (UUT) to an [unmanaged multiport PoE switch](#). Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet as the FXO Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add each channel of the UUT to an audio group. On the Group Membership Tab for each channel add auto DTMF digits to the group of which the UUT is a member. Update/reset the UUT to achieve normal status. Make certain that all conflicts are cleared.

Connect your lineman's handset in monitor mode and *connect the tip/ring of the analog station port or intercom/paging system that will be used with the UUT* to the UUT's line output.

Go off-hook with your lineman's handset to verify that you receive dial tone and/or battery feed voltage from the tip/ring of the analog station port or intercom/paging system.

Place your lineman's handset back to monitor mode.

Use the Communications/Send sample page utility of the VIP-102B to send sample audio to the group that includes the UUT. You should hear the auto DTMF digits via the lineman's handset.

Retest for each line output of the UUT.

These are basic tests. If using Session Initiation Protocol (SIP) for access, then following the basic test, the UUT should be relocated to, and properly configured to work on, a network segment that is routable to the telephone system VLAN. Properly configure the SIP Tab(s) of the product being tested and access the UUT via a SIP telephone. Refer to the SIP (Session Initiation Protocol) Troubles section for more information.

Audio Gateway channels used as an output

Connect the Audio Gateway under test (UUT) to an unmanaged multiport PoE switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet as the Audio Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add each channel of the UUT to an audio group.

Program each relay output for "Activate on Call" and map it to a channel.

Update/reset the UUT to achieve normal status. Make certain that all conflicts are cleared.

Connect your lineman's handset in monitor mode to the UUT's audio output.

Connect your voltmeter probes to the relay output connections and verify that there is no dc or ac voltage present.

Test your Ohmmeter by touching the metal probes together. The display should indicate a short circuit (approximately zero Ohms). When the metal probes are not connected to anything, the display should indicate an open circuit.

Connect your Ohmmeter probes to one of the relay outputs that is mapped to first channel audio output that you will be testing.

Use the Communications/Send sample page utility of the VIP-102B to send sample audio to the group that includes the UUT. You should hear the sample audio in the lineman's handset and the Ohmmeter should indicate a short circuit.

Retest for each audio and relay output of the UUT.

These are basic tests. If using Session Initiation Protocol (SIP) for access, then following the basic test, the UUT should be relocated to, and properly configured to work on, a network segment that is routable to the telephone system VLAN. Properly configure the SIP Tab(s) of the product being tested and access the UUT via a SIP telephone. Refer to the SIP (Session Initiation Protocol) Troubles section for more information.

Audio Gateway channels used as an input

Connect the Audio Gateway under test (UUT) and a known working IP speaker to an unmanaged multiport PoE switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT and the IP speaker. Set the PC IP address to be on the same subnet as the Audio Gateway (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

If the IP speaker is on a different subnet than the UUT, then reassign its IP address so the 2 endpoints are on the same subnet (i.e. 192.168.6.201)

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add the IP speaker to an audio group. Do not add the UUT channel(s) to the audio group. On the VIP-102B's Channel Tabs for the IP speaker and each UUT, change the output volume to -15.

There are 2 ways to initiate streaming of the audio connected to an Audio Gateway input. One is using Voice Operated Switching (VOX), and the other is by applying a contact closure to the Gateway input.

If testing VOX operation:

On each Channel Tab of the UUT, check "use VOX activation" and enter the audio group's access code in the auto destination field. On each Input Tab of the UUT change input function to "not active".

Using the VIP-102B, update/reset to achieve normal status of both endpoints. Make certain that all conflicts are cleared.

Disconnect all cabling from the audio input(s) of the UUT.

Verify that the line level audio source you will be using for test is working by monitoring the output with your inductive amplifier or lineman's handset.

Connect the output of your line level audio source to one of the channel Audio Inputs of the UUT.

You should hear the audio through the IP speaker. If not, increase the audio level of your line level audio source.

Retest for all line inputs of the UUT.

If testing contact closure activation:

On each Channel Tab of the UUT, uncheck "use VOX activation" and enter the audio group's access code in the auto destination field. On each Input Tab of the UUT change input function to "Audio On/Audio Off".

Using the VIP-102B, update/reset to achieve normal status of both endpoints. Make certain that all conflicts are cleared.

Disconnect all cabling from the audio input(s) of the UUT.

Verify that the line level audio source you will be using for test is working by monitoring the output with your inductive amplifier or lineman's handset.

Connect the output of your line level audio source to one of the channel Audio Inputs of the UUT.

Apply a short circuit to the Contact Closure Input of the same channel.

You should hear the audio through the IP speaker. If not, increase the audio level of your line level audio source.

Retest for all line inputs of the UUT.

Application Servers

Connect the Application Server under test (UUT) and a known working IP speaker to an unmanaged multiport PoE switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT and the IP speaker. Set the PC IP address to be on the same subnet as the Application Server (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

If the IP speaker is on a different subnet than the UUT, then reassign its IP address so the 2 endpoints are on the same subnet (i.e. 192.168.6.201)

Rescan the switch. You should now be able to receive group membership information from the UUT and the IP speaker.

Using the VIP-102B, add the IP speaker to an audio group. On the VIP-102B's Channel Tab for the IP speaker, change the output volume to -15. Update/reset to achieve normal status of both endpoints. Make certain that all conflicts are cleared.

Browse into the Application Server by entering its IP address into a browser (Firefox preferred). Log into the server, go to the quick page menu and send an audio quick page to the group that includes the IP speaker. You should hear the audio from the speaker.

The Application Servers have log files that are typically useful for evaluating activity and troubles.

Telephone Paging Servers

Connect the Telephone Paging Server under test (UUT) to an unmanaged multiport switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet as the Telephone Paging Server (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Browse into the Telephone Paging Server by entering its IP address into a browser (Firefox preferred). Log into the server, the Telephone Paging Server has a log file that is typically useful for evaluating activity and troubles.

Further evaluation requires moving the UUT to the actual network and following the installation manual to establish communications with the VoIP telephone system.

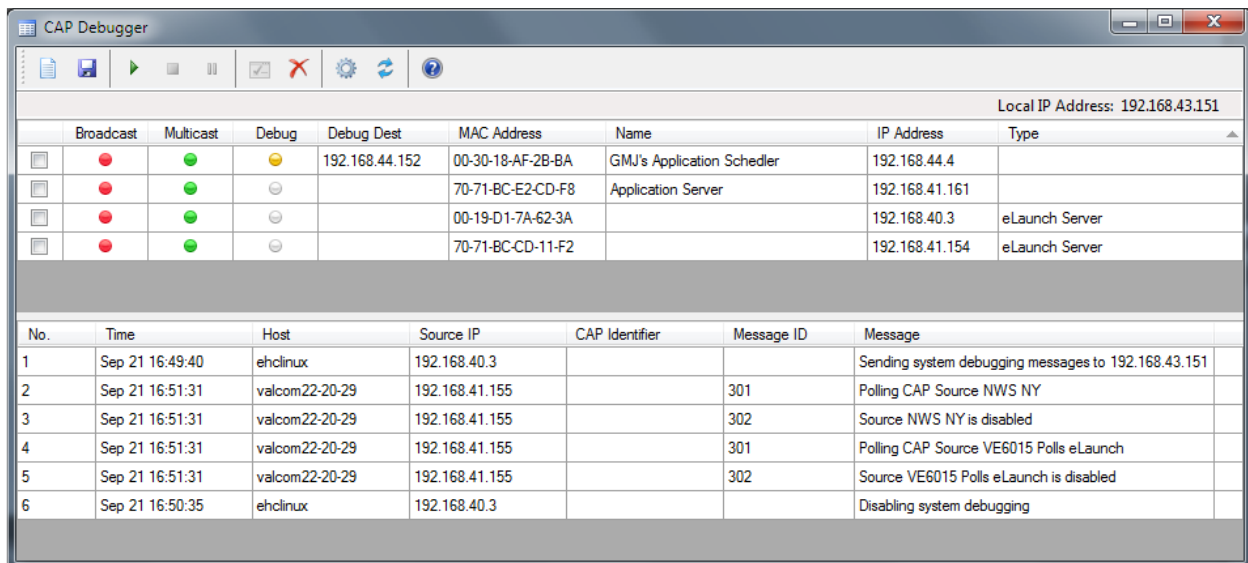See the Product Specific Troubles/Tips section for more.

eLaunch Server

Connect the eLaunch Server under test (UUT) to an unmanaged multiport switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE switch.

Scan the switch with the VIP-102B to determine the IP address of the UUT. Set the PC IP address to be on the same subnet as the eLaunch Server (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Browse into the eLaunch Server by entering its IP address into a browser (Firefox preferred). Log into the server, Further evaluation requires configuring the eLaunch Server on the actual network and using the CAP Debugger Tool to view the progress of CAP alerts as they are generated.

The CAP Debugger Tool

The CAP Debugger, automatically installed with the VIP-102B, is used to trace the progress of an alert generated by the eLaunch system and handled by the Application Server Pro devices.

	Broadcast	Multicast	Debug	Debug Dest	MAC Address	Name	IP Address	Type
☐	🔴	🟢	🟡	192.168.44.152	00-30-18-AF-2B-BA	GMJ's Application Schedler	192.168.44.4	
☐	🔴	🟢	⚪		70-71-BC-E2-CD-F8	Application Server	192.168.41.161	
☐	🔴	🟢	⚪		00-19-D1-7A-62-3A		192.168.40.3	eLaunch Server
☐	🔴	🟢	⚪		70-71-BC-CD-11-F2		192.168.41.154	eLaunch Server

Local IP Address: 192.168.43.151

No.	Time	Host	Source IP	CAP Identifier	Message ID	Message
1	Sep 21 16:49:40	ehclinux	192.168.40.3			Sending system debugging messages to 192.168.43.151
2	Sep 21 16:51:31	valcom22-20-29	192.168.41.155		301	Polling CAP Source NWS NY
3	Sep 21 16:51:31	valcom22-20-29	192.168.41.155		302	Source NWS NY is disabled
4	Sep 21 16:51:31	valcom22-20-29	192.168.41.155		301	Polling CAP Source VE6015 Polls eLaunch
5	Sep 21 16:51:31	valcom22-20-29	192.168.41.155		302	Source VE6015 Polls eLaunch is disabled
6	Sep 21 16:50:35	ehclinux	192.168.40.3			Disabling system debugging

The top section of the tool will list all of the eLaunch and Application Server Pro devices that have been detected on the network from the beacons they are sending. The bottom section will display syslog messages from selected devices as alerts are generated.

To use the tool, put checks beside all the devices that you wish to debug and click the Start button.

The selected devices will enter debugging mode as indicated by the Debug status indicator. The destination of their debug messages will be set to the IP address of the local PC.

Debug	Debug Dest
🟡	192.168.44.152
⚪	
🟢	192.168.43.151
⚪	

A green indicator tells us that debug messages are being sent to our local PC. A yellow indicator tells us that the device is sending debug messages to some other PC on the network. A white indicator tells us that the device is not currently in debug mode.

At this point, an alert can be generated from the eLaunch system and debug messages should appear in the lower section of the tool. If both an eLaunch system and an Application Server Pro were selected for debugging, messages should be seen from both devices as the alert is generated and then received and acted upon.

When debugging is finished, the Stop button can be pressed which will cause any device that is sending debug messages to the local PC to exit debug mode.


the Stop ALL Debugging button stops ALL devices that are sending debug messages, even those that are sending to other destinations:

The Pause button can be used to temporarily halt receiving messages while debugging without actually telling the devices to exit debugging mode:
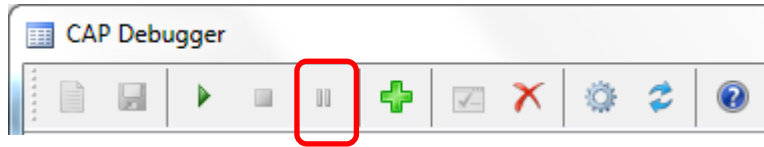
While a capture is running, the Change Selected Devices button can be used to modify the devices that should be sending messages without having the stop the current capture first. Simply adjust the necessary check marks beside the desired devices and click the button to change the devices that will be sending debugging messages to the local PC:
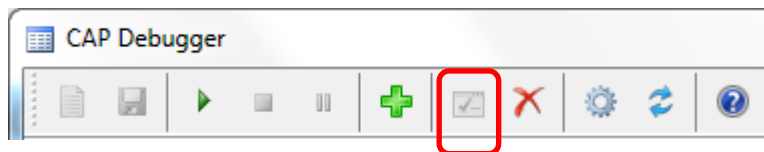
The New button can be used to clear the list of messages received and start a new capture using the selected devices:
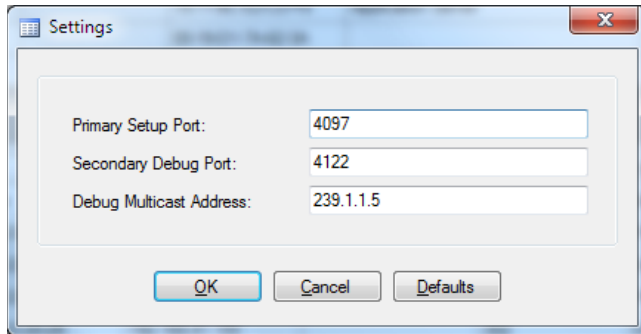

The Save button can be used to save the messages from the current capture to a text file:

The Settings button can be used to change the address and ports that the tool is communicating on if these settings have been modified in the actual VIP devices using the VIP-102B tool:

The Restart Communications button can be used to restart communications if some error occurred during startup and communications could not be initialized. This could be caused by problems such as other tools running that are already using the specified ports or by network connectivity issues:

If there were issues on startup, the button may appear with a red background to indicate that there were problems, and the other buttons might be disabled due to the restricted functionality until the problem is resolved.

Relay Outputs

Remove all connections from the relay output.

Connect your voltmeter probes to the relay output connections and verify that there is no dc or ac voltage present.

Test your Ohmmeter by touching the metal probes together. The display should indicate a short circuit (approximately zero Ohms). When the metal probes are not connected to anything, the display should indicate an open circuit.

Connect your Ohmmeter probes to the relay output connections.

If the relay contact is a normally open relay contact (normally meaning not energized or idle) then the Ohm meter will indicate an open circuit.

If the relay contact is a normally closed relay contact (normally meaning not energized or idle) then the Ohm meter will indicate a short circuit (approximately zero Ohms).

The resistance of the relay output connections should change state when the relay is activated. Normally open contacts should close (short circuit) and normally closed contacts should open.

Switch Inputs

Remove all wiring from your switch input. Set your voltmeter to a range suitable for the expected voltage of the switch input. For example, if the switch input is part of equipment being powered by 24vdc, then you would expect the input voltage to be 24vdc or less, etc. If in doubt, always start at a higher voltage setting and work your way down to the correct setting.

Use your voltmeter to verify the presence of dc or ac voltage on the switch input. Short the switch input. Verify that the dc or ac voltage is now zero volts. Note that some switch inputs are actually opto-coupler inputs and rely upon an external current and will not have dc voltage present when the field wiring is disconnected.

Activation Closures/Switches

Normally open switches, pushbuttons, call switches and activation closures (switch for this discussion) connected to a system input may be tested in a couple of ways. The first is to disconnect the switch from any equipment and place an Ohmmeter across the unterminated switch contacts. When the switch is activated, the meter should indicate close to zero Ohms. When the switch is not activated, then the meter should indicate an open circuit.

The other test actually verifies operation of both the switch and the equipment input. In this test, do not disconnect the switch from the equipment. Set your voltmeter to a range suitable for the expected voltage of the switch input. For example, if the switch input is part of equipment being powered by 24vdc, then you would expect the input voltage to be 24vdc or less, etc. If in doubt, always start at a higher voltage setting and work your way down to the correct setting

Measure the voltage across the switch input while the switch is activated and when it is deactivated. When the switch is activated, the input voltage should drop significantly.

If you never measure any voltage, then refer to section above on testing switch inputs.

Remember that wire connecting the switch and input is also a variable, so temporarily moving the switch close to the input, or assessing the switch with the Ohmmeter without any connected field wiring may be necessary.

If the switch is normally closed, instead of normally open, then your results will be backwards.

DC Power Supplies

Remove all wiring from your power supply outputs. Use your voltmeter to verify the presence of the rated dc voltage on the supply output. If the voltage drops significantly once the field wiring is reconnected, then the load is excessive or the field wiring will require further investigation.

Line level audio outputs

Remove all wiring from your line level audio output. Initiate audio through the line level output. Use your lineman's handset in monitor mode to verify the presence of the audio. If the audio level drops significantly once the field wiring is reconnected, then the load is excessive or the field wiring will require further investigation.

Talkback speaker level audio outputs

Remove all wiring from your audio output. Initiate audio through the audio output. Use your lineman's handset in monitor mode to verify the presence of the audio. If the audio level drops significantly once the field wiring is reconnected, then the load is excessive or the field wiring will require further investigation.

IP LED Signs

Connect the LED Sign under test (UUT) to an unmanaged multiport PoE+ switch. Connect a PC that has the VIP-102B IP Solutions Setup Tool installed to another port on the PoE+ switch.

Scan the switch with the VIP-102B to determine the IP address of the LED Sign. Set the PC IP address to be on the same subnet as the LED Sign (i.e. if the IP address of the UUT is 192.168.6.203, then change the PC IP address to 192.168.6.200).

Rescan the switch. You should now be able to receive group membership information from the UUT.

Using the VIP-102B, add the LED Sign to an audio group. Update/reset to achieve normal status. Make certain that all conflicts are cleared.

Use the VIP-102B Communications /Send Sample Page to send text to the audio group that includes the LED Sign. The sign should display the text. Be certain to check the box next to the "Text Message" field.
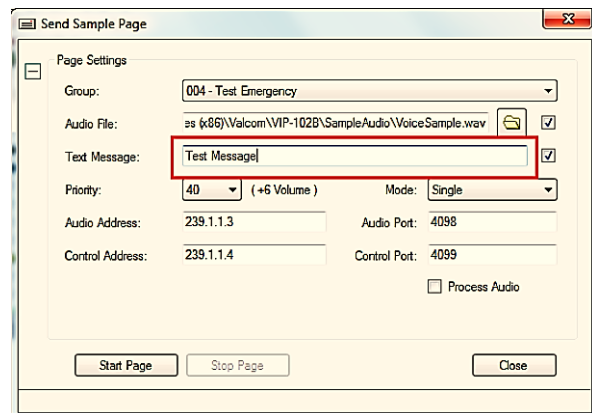
Trouble Scenarios

Hot Equipment

Valcom equipment should not feel uncomfortably hot to the touch. If this occurs remove power from the unit, remove all field wiring, reapply power and see if the heat condition returns with no field wiring. If it does, then return the Valcom equipment for repair. If the equipment feels cool without field wiring, then assess your field wiring for grounds, short circuits and excessive load before reconnection.

Products not operating properly

If you have a Valcom product that is not operating as it should then double check all connections. Make certain that they are correct, and secure. Verify all switch settings, device network settings, programming, volume adjustments, etc., and then cycle power to reboot the device. If you have another identical product that works properly, swapping the location and programming of the 2 may help to determine if the trouble is install related or equipment related. If the trouble stays with the location, then it is most likely not equipment related. If an individual input or output (I/O) of a piece of equipment is experiencing trouble, but others are not, evaluate whatever is connected to that I/O as a possible cause. Swap cabling with a working I/O to see if the trouble follows the I/O or cabling. If testing indicates that the trouble is equipment related, then contact technical support.

In the case of PoE powered VoIP equipment, assess the suspect product by isolating it to an unmanaged PoE/PoE+ test switch. If it behaves properly on the unmanaged PoE/PoE+ test switch, the trouble may be related to network settings, PoE power budget, or PoE/PoE+ switch firmware. We have experienced sites with multiples of the same model endpoint where specific endpoints exhibit troubles when connected to the system switch but work perfectly on an unmanaged PoE test switch, yet other identical endpoints do not experience any issues. In these instances, simply upgrading the system PoE switch firmware rectified the trouble.

Products that require PoE+ that begin to exhibit trouble may not be receiving full power from the PoE+ switch. It is possible for PoE+ switch ports to revert to low power delivery due to many different conditions unrelated to the Valcom device. For example, if a PoE+ switch requires multiple power supplies to provide full PoE+ power to all ports, and one of the Power Supplies fails, then the switch may revert to limited power to all ports. The behavior of the PoE+ switch is dependent on the manufacturer.

Also refer to:

The importance of multicast in Valcom VoIP systems

No group/all call audio from speaker

This may be a cabling issue, a network issue or a hardware issue. In IP based systems this is a classic example of improperly configured multicast (refer to the IP6000 Initial Setup Procedure, the Valcom VoIP Initial Setup Procedure or the Testing Multicast section of this document). A classic example of improperly configured multicast has the following characteristics:

1. IP speakers will produce an audible click during a group announcement in response to successful reception of the control multicast address.
2. Following this "click" the page audio will not broadcast.

If you use the Paging Diagnostics of the VIP-102B IP Solutions Setup Tool to monitor the VLAN, you will notice that the Page Start is present, but no data is received (Refer to the following showing a successful announcement from HS Analog Phone followed by an unsuccessful announcement from HS IP Phone Gateway:

Date	Time	Action	Group	Source Name	Source IP	Priority	Data Received
07/09/2020	01:44:18 PM	Page Start	1000	HS Analog Phone	172.16.30.6	10	473
07/09/2020	01:44:28 PM	Page Stop	1000	HS Analog Phone	172.16.30.6	10	473
07/09/2020	01:44:53 PM	Page Start	1000	HS IP Phone Gateway	172.16.30.3	10	
07/09/2020	01:45:02 PM	Page Stop	1000	HS IP Phone Gateway	172.16.30.3	10	

This is indicative of the control multicast address being properly routed to the PC hosting the VIP-102B while the audio multicast is not.

Also refer to:

The importance of multicast in Valcom VoIP systems

If one-way audio, (group announcements, tones, music initiation, etc.) require **multiple initiation attempts** after being idle for some period of time, then see the Intermittent Troubles section of this document.

When troubleshooting analog speaker circuits, trace through the speaker circuit with your lineman's handset as described previously. It's possible to be able to successfully call an individual audio output, however, have that same output fail during group/all call announcements. Disconnect the speaker wiring from the audio output and connect your lineman's handset (in monitor (hi-z) mode) to the audio output. Try calling the individual audio output as part of a group and by its own unique access code. If both successfully page into the lineman's handset, then you have a wiring issue.

Low All Call/Tone Volume

Some analog systems have a preset, non-adjustable all call volume. In these systems, this preset applies to all call, group announcements and tones. The intent of this design is to facilitate future maintenance. On analog systems designed for retrofits, an all call volume control is provided to match the level of the replaced system. Since the all call volume and tones volume are associated with each other, complaints of low all call or low tones should be analyzed to determine if they are both low. The all call volume on retrofit system cards will increase or decrease both. If only one is low, contact technical support.

In the case of IP systems, many volume controls need to be considered when troubleshooting speaker volumes. There are separate volume controls for groups based upon priority, there are individual channel volume controls, there are FXS gateway volume controls and there are "per event" server volume controls.

Are there volume controls on system amplifiers or in line with speaker circuits that may need to be adjusted?

Low/No Talkback Volume

Some analog systems have a preset, speaker-to-phone volume control.

In the case of IP systems, many volume controls need to be considered when troubleshooting Talkback speaker volumes. There are separate volume controls for individual channel volume controls and there are FXS gateway input volume controls.

Also refer to:

The importance of multicast in Valcom VoIP systems

No sound from speaker

This may be a cabling issue, a network issue, or a hardware issue. For IP speakers, check programming, volume control, PoE port stability device network settings (missing or incorrect gateway?) and facility network settings (refer to the IP6000 Initial Setup Procedure or the Valcom VoIP Initial Setup Procedure).

Also refer to:

The importance of multicast in Valcom VoIP systems

When troubleshooting analog speaker circuits, trace through the speaker circuit with your lineman's handset as described previously. Disconnect the speaker wiring from the audio output and connect your lineman's handset (in monitor (hi-z) mode) to the audio output.

Try calling the individual audio output as part of a group and by its own unique access code. If both successfully page into the lineman's handset, then you have a wiring issue or suspect speaker.

Checking Valcom Self Amplified speakers or horns is incredibly easy. First, verify that the integrated volume control is <u>not</u> turned all the way down. Next, place your volt ohm meter on the voltage input of the speaker or horn. Simultaneously, attach your lineman's handset in monitor (hi-z) mode to the audio input of the speaker or horn. Have someone initiate a test announcement*. With the volume control turned up, if you hear audio on the audio input through your lineman's handset and measure adequate voltage on the voltage input then the speaker or horn should broadcast sound. If not, replace it. If the dc voltage at idle is adequate, but significantly drops (drops more than 6 VDC directly at the speaker or horn power input) during the announcement, then either the power supply or wiring is undersized for the speaker wire run. Refer to the following Power Pair Run Chart.

*Loud audio from speakers, especially common when in proximity, can cause hearing damage.

Power Pair Wire Run								
Number of Speakers/Horns Per Power Run					Power Run Wire Length in Feet (meters)			
1 VPU* Speakers	4 VPU* Speakers	Flex Horns	5-Watt Horns	15/30-Watt Horns	24 AWG	22 AWG	20 AWG	18 AWG
4	1	-	-	-	1000' (304 m)	1600' (487 m)	2500' (762 m)	4000' (1219 m)
7	2	1	1	-	500' (152 m)	800' (243 m)	1280' (390 m)	2025' (617 m)
15	4	2	2	-	250' (76 m)	400' (122 m)	640' (195 m)	1010' (308 m)
30	8	4	4	1	125' (38 m)	200' (61 m)	320' (98 m)	500' (152 m)

*VPU = Valcom Power Unit

Low/No Speaker Volume (all call or individual zone)

Once you've eliminated cabling issues and verified that the speaker is functional (refer to the "No Sound From Speaker" section), check all system volume controls, including those that may be in line with the speaker (wall mount volume controls, call switch volume controls, V-LPTs, amplifier volume controls, etc.)

Low audio from a speaker, if unrelated to volume control setting or a low output from the audio source, is typically a cable length or connection issue (for example, a bad connection on a punch down block). Prove it by temporarily moving the speaker in

question into the equipment room and connecting it to the source with a short piece of cable. If the low audio condition is alleviated, then the cabling/connectivity are suspect.

In the case of self-amplified or IP speakers, inadequate power (PoE or power supply) may be the culprit. If speaker amplifiers are starved for power, they can't provide adequate audio power. Another common symptom of inadequate power is distorted audio, but not always.

Low audio may also be caused by user error. Users holding the originating microphone or telephone too far away or speaking too softly into the originating microphone or telephone is a common issue.

In the case of IP systems, if not a PoE issue, many volume controls need to be considered when troubleshooting speaker volumes. There are separate volume controls for groups based upon priority, there are individual channel volume controls, there are FXS gateway volume controls and there are "per event" server volume controls.

Lastly, improperly sized surge protectors have been known to restrict signal magnitude thus restricting power delivery to speakers or horns.

Also refer to:

The importance of multicast in Valcom VoIP systems

Cannot access individual multizone intercom/PA channels
Most VoIP gateways and endpoints have a "phone book" that contains information about the IP system's *statically assigned* IP address/channel dial code pairs. This phone book is populated via the VIP-102B. These VoIP Gateways and endpoints use the phone book entries to locate other channels in the system.

The VECPU6, VECPU6-EXP, VENSCA, VE8006R, and any VoIP endpoints using DHCP, find unknown channel dial code destinations via multicast. These dial code routes are dynamically added to the phone book as found. If an entry in the phone book fails, perhaps due to the DHCP assigned address changing, then a new multicast query is sent to obtain current information and the phone book is updated.

Regardless of the use of DHCP or static addressing, the use of groups, such as all call, require properly configured multicast.

If multicast is not properly implemented when using the VECPU6, VECPU6-EXP, VENSCA, VE8006R, and any VoIP endpoints using DHCP, there may be issues with calling both individual VoIP gateway and speaker channels and/or accessing groups.

As far as analog system, although extremely rare, systems that utilize 40-pin ribbon cables to connect a CPU to station cards may experience issues with calling individual channels if the ribbon cable is not seated properly or is damaged. If there is a pattern to successful/unsuccessful channel call attempts such as every 3rd attempt fails, or if 2 calls are in progress, the third will fail, or the second call is only successful if the first call has concurrently failed, then the ribbon cable should be replaced or examined for proper seating.

Useful Info:

VECPU5 has 8 analog paths, one of which is dedicated to the confirmation tone, if enabled. It cycles through the 7 or 8 available paths sequentially.

VECPU6/VECPU6-EXP/VENSCA have 4 one-way audio paths and 4 handsfree audio paths. It always uses the first available path.

If you suspect a ribbon cable issue on VECPU6/VECPU6-EXP/VENSCA, you'll need to generate one or more simultaneous calls to verify the suspicion.

Poor Quality Background Music

If the music sounds distorted, then the music source may be turned up too loud. Try turning the music source down.

Most music sources have left and right channel stereo outputs. Stereo audio, by definition, cannot be supported by public address systems. Therefore, the source's left and right channels must be mixed together to form a single monophonic output. It's not unusual to visit sites where only one of the stereo channels is connected to the PA system or where both stereo channels are connected to one music input. Either will result in poor music quality and possible damage to the audio source.

Valcom's Remote Input Module ([V-9130-W](#)) provides an inexpensive means to properly mix stereo audio to monophonic audio.

Remember that background music, as the name implies, is low level audio intended for subliminal recognition. High fidelity is not often a requirement.

Foreground music, music intended to be blatantly obvious to the listener, often requires high fidelity sources and speakers. This equipment is typically costlier, and less versatile than audio paging equipment.

IP systems that produce high quality foreground music use a significant amount of bandwidth and require a network capable of providing that bandwidth. Valcom IP systems are optimized for voice and should not be promoted for high fidelity music distribution.

Noise/Feedback

There are 2 common types of noise that may occur in audio systems, 60 Hz hum and feedback squeal. Hum is typically related to either:

> a) The physical location of the installed equipment and its proximity to other equipment and/or wiring

> b) A mismatch in ground potential between a device sending audio and a device receiving audio

> c) "Dirty" ac voltage (we've "cured" hum problems simply by using a different ac outlet for our equipment)

> d) Lightly loaded dc power supplies

Use a lineman's handset in monitor mode to listen to 24- or 48-volt DC power supply outputs used to power audio circuits. These DC outputs should be quiet.

If noisy power supplies are suspected of inducing noise into a system, temporarily substitute the supplies with DC batteries (2 9-volt batteries connected in series will power most 24-volt speakers and circuits). If the batteries quiet the noise, then the power supplies are the likely noise source.

The way to determine if the hum is physical location related is to move the equipment in question to a different environment and test. The author has witnessed equipment that exhibited hum due to high voltage transformers of the other side of the equipment wall and also from high voltage cabling routed near the Valcom equipment.

a) The intent of using UTP wiring with a differential audio input is to reject noise. Not maintaining the integrity of the twist in UTP renders it ineffective. The twist in UTP pairs should be maintained right up to the termination points.

b) Ac power wiring and audio signal input wiring should be routed separately whenever possible. The electromagnetic field surrounding conductors carrying ac voltage increases in strength following increases in current. This electromagnetic field induces crosstalk into surrounding conductors.

c) Improper grounding can cause 60 Hz hum.

Mismatches in ground potential between a device sending audio and a device receiving audio are easily remedied by routing the line level audio through isolation transformers. Valcom VMT-2 1:1 isolation transformers have solved many hum problems and should be standard issue to all audio technicians.

"Dirty" ac voltage is ac voltage with a significant amount of electrical noise superimposed. The condition may be proved by powering the equipment experiencing the hum by an adequately charged, unplugged UPS. A licensed electrician should be contacted to rectify "dirty" ac issues.

Lightly loaded power dc supplies may require some additional load for internal filtering to work properly.

Hum from microphone-initiated announcements is often the result of connecting the microphone via UTP cabling. Microphones connections should always utilize good quality microphone cable. (Belden 8424 or equal)

Improper signal to noise ratio often results in noise troubles. For example, when providing an excessively low audio signal to an amplifier, the ratio of noise to desired audio signal in the cabling can become significant. To have acceptable broadcast levels for the desired signal, the amplifier gain must be increased, which of course increases the volume of the noise as well.

By increasing the desired signal into the amplifier, if the induced noise level remains constant, the amplifier gain can be lowered thus lowering the noise while still providing adequate broadcast levels for the desired signal.

As is true of many things, audio amplifiers, speakers and horns are not typically intended to operate at full rated capacity/volume 100% of the time. A system design requiring audio devices to continuously operate at maximum volume is a poor design and much more likely to experience audio quality troubles and premature equipment failure.

Squeal

A high-pitched squeal through paging speakers and horns is often the result of acoustic feedback. This results when amplified audio loops back into the originating microphone or telephone increasing its gain with each successive loop until the system amplifiers reach saturation. This results in the high-pitched squeal. Remedies include delaying the announcements through feedback eliminators, lowering or eliminating the sound level near the originating microphone or telephone and using noise cancelling microphones or telephones. System speaker grouping can often be used to create separate all call groups for each paging initiation area eliminating speakers in that area during general announcements.

Noisy Talkback Speaker to Phone Audio

Talkback speakers do not discriminate between voice and room noise. If the area where the speaker is located is noisy, then the talkback audio will also be noisy. Secondly, the audio signal level from analog talkback speakers is similar to the audio signal level produced by a microphone. This signal is greatly amplified by the receiving equipment. In most cases, common mode rejection nulls low level noise induced into cabling.

If the received talkback signal is noisy, be certain to use the manufacturer suggested cable type and distance guidelines and avoid:

a) splitting twisted pairs
b) using extra pairs in talkback speaker cables for noisy signals

c) routing wiring near light ballasts or other equipment with strong electromagnetic fields or wire conductors carrying high level ac signals

d) installing talkback speaker locations directly in the path of active fans or HVAC vents

Low Audio from Amplifier

Are there volume controls on system amplifiers or in line with amplifiers/speakers that may need to be adjusted?

To produce adequate output levels, amplifiers require adequate input levels. The signal level on an amplifier's input that is required to achieve rated output is referred to as its "sensitivity". It's equally important for the source's output impedance to be <= the amplifier's input impedance. Maximum signal transfer (amplifier input) occurs when source impedance is zero and input impedance is infinite. Maximum POWER transfer (amplifier output) occurs when output impedance is equal to load impedance.

Excessive speaker load is another possible cause of low amplifier audio output. All amplifiers are designed with a maximum speaker load in mind. Disconnect the speaker line(s) from the amplifier and check the load with an impedance meter.

In systems using old fashioned 25/70.7/100-volt amplifiers, undersized wiring resulting in too much line loss will cause low audio. Refer to the *Low sound from speaker (all audio)* section of this document.

Low or no sound from an audio source

If an audio source, like a page port or music source appears to produce audio when tested with a lineman's handset but produces low or no audio when connected to a system input, it may be that the one or both wire conductors between the source and the system input is not terminated properly. Sometimes connections look proper when they are not. Start over, check all connectors and re terminate the source.
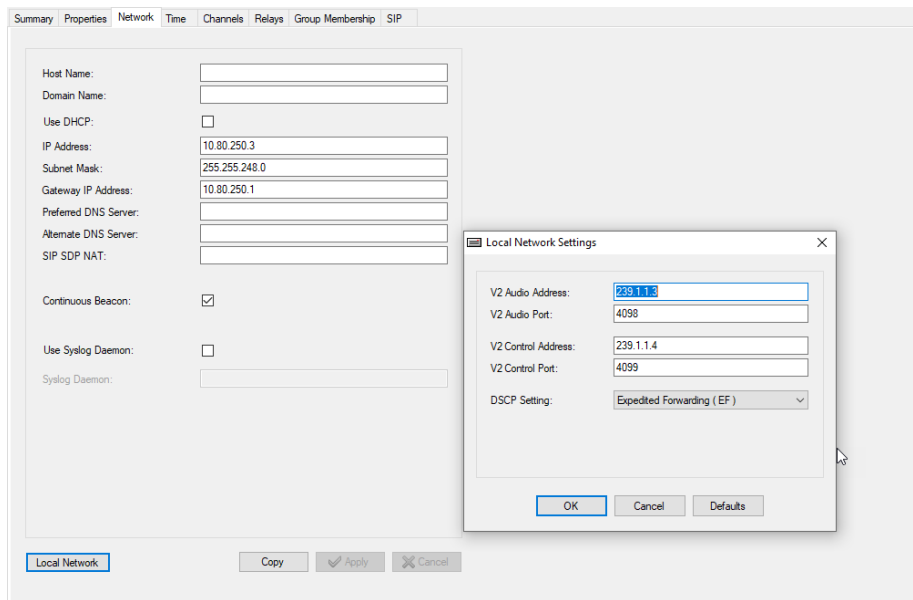
Also refer to:

The importance of multicast in Valcom VoIP systems

Poor Quality VoIP Audio

Improper multicast configuration, incorrect/mismatched DSCP setting (formerly QoS), poorly terminated RJ45s, 110 blocks and/or variances in network traffic can cause audio quality issues. Heavy network traffic can result in broken or missed audio and unreliable system operation. While VLANs do not increase the available bandwidth of a network, they can restrict traffic only to ports where it is intended. Therefore, Valcom always highly recommends a dedicated VLAN, or a VLAN shared with the telephone system. You can test for network issues by temporarily moving the Valcom VoIP devices to a dedicated, non-configured multiport PoE or PoE+ switch for testing.

Check with your network administrator for the proper DSCP setting.

Broken, Intermittent, Garbled or Choppy Audio

For analog speaker circuits, intermittent connections, or overloaded power supplies/amplifiers/PoE switches/cable may result in broken, intermittent or choppy audio. Use a lineman's handset in monitor mode to determine where the choppy audio first occurs in the system (at the source, after the amplifier/zone controller, etc.).

Test for overloaded supplies/amplifiers/PoE switches/cable by testing with most or part of the load (speakers, horns, etc.) temporarily removed.

For networked equipment, improper multicast configuration, incorrect/mismatched DSCP setting (formerly QoS), poorly terminated RJ45s, 110 blocks and/or variances in network traffic can cause issues. Heavy network traffic can result in broken or missed audio and unreliable system operation. While VLANs do not increase the available bandwidth of a

network, they can restrict traffic only to ports where it is intended. Therefore, Valcom always highly recommends installation on a dedicated VLAN, or a VLAN shared with the telephone system.

In cases where the originating and receiving endpoints are not dependent upon other systems and are configured with unique static IP addresses in one subnet, they may easily be moved from the facility's LAN to a non-configured multiport PoE or PoE+ switch for testing. If the equipment works properly while connected to the dedicated test switch, but does not work properly on the facility network, then facility network issues or improper configuration are highly probable.

For example, if you experience trouble making an announcement through one or more IP speakers from a telephone connected to an FXS gateway, and the speakers and the FXS gateway all have static IP addresses in the same subnet, simply move one or more of the speakers and the FXS gateway to the test switch. If the trouble clears when the equipment is connected to the test switch, the facility LAN setup will require evaluation.

If using VERCA cards and all network requirements have been met, refer to the echo cancellation procedure in the manual.

Talkback systems may experience broken audio during intercom (talkback) conversations if the originator speaks too slowly or softly. Some products offer adjustment of audio switching sensitivity (attack) and telephone release (hold) time.

Testing Talkback Switching

Talkback speaker systems allow 2-way communication between a speaker location and a telephone. In most cases, the speaker cone acts as both a broadcast speaker and as a microphone. The number of talkback speakers on a circuit is often limited to 2 to minimize noise interference.

Many intercom systems are half duplex, meaning only one person can talk at a time. The individual on the telephone end of the conversation controls the audio. When they speak, the audio direction switches to deliver the phone's audio to the speaker. The level of audio required to invoke this switch is known as the "attack" level.

After the individual at the phone end stops speaking for a time, the audio direction will switch to allow the audio generated at the speaker to be transmitted to the phone. This time is known as the release value and is measured in milliseconds.

Phones located in noisy areas may experience trouble hearing the speaker audio as the background noise will "trick" the system into attack mode. The result will be "choppy"

speaker to phone audio. Increasing the attack level, if possible, will help to alleviate this choppy audio condition.

If audio from the phone to the speaker is "clipped", then lowering the attack level, or increasing the release time may help.

With the attack level at its lowest setting, even normal breathing would keep the phone-to-speaker audio active.

With the attack level at its highest setting, the phone would not be able to gain control and the speaker-to-phone audio would be ever present.

A couple of good tests for talkback audio include:

a) have an individual at the speaker say random short numbers which are then repeated by the individual at the phone.

b) having the individual at the phone read a paragraph, the audio should not switch unless there are long pauses in the recital.
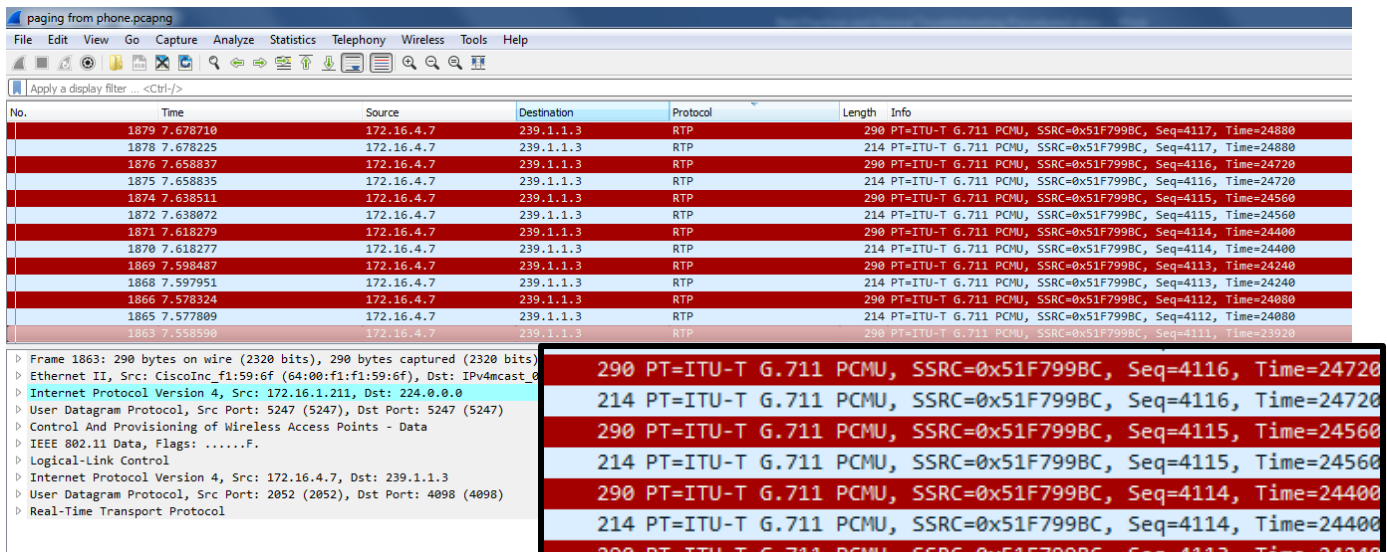
Distorted sound from speaker

Distorted sound from a speaker is almost always the result of overdriving the speaker, or overdriving the device sourcing the audio to the speaker. Overdriving the input of an amplifier or self-amplified speaker will result in clipped, and therefore distorted, speaker audio.

For Valcom Self amplified speakers, exceeding the Valcom wire gauge guide or not providing enough Valcom Power units (inadequately sized power supplies/cabling) may also result in distorted audio. In this scenario, the distortion is typically most apparent on speakers furthest from the power supply. In many cases, you can test for inadequate power or excessive cable loss by reducing speaker volumes (thus lessening the load) and checking to see if the clarity of the speaker audio improves.

As stated previously, audio amplifiers, speakers and horns are not typically intended to operate at full rated capacity/volume 100% of the time. A system design requiring audio devices to continuously operate at maximum volume is a poor design and much more likely to experience audio quality troubles and premature equipment failure.

In VoIP systems, improper network setup can result in packet duplication such that IP speakers play the same audio twice or more. The resulting sound is distorted, or perhaps even echoic. In the Wireshark screen capture below notice that there are multiple RTP packets with the same SSRC and Seq numbers. These are duplicate packets that were being introduced by a device with IP address 172.16.1.211. See the troubleshooting process here.

Lastly, improperly sized surge protectors have been known to restrict signal magnitude resulting in clipped audio signal delivery to speakers or horns. This clipped signal results in distorted sound.

Low Microphone Audio

If audio originating from a microphone is low and/or of inferior quality, check to make certain that the microphone is the correct type and has the correct output impedance for your system. Note that some microphones have selectable output impedance.

Errant Audio

If music, tones or voice are heard in speaker areas other than those to which it is directed, then the issue could be crosstalk related and you might review the section of this guide dedicated to troubleshooting crosstalk. However, if bell tones are being broadcast to incorrect areas in an Application Server based system, the culprit could be weak filters in the text monitor used to invoke Play Lists. Oftentimes, text monitors and Play Lists are used to add manual, dial code activated bells to IP systems. The bells are triggered by a group dial code preceded by syslog information indicating that is was initiated from a specific phone or phones. If the filter simply contains the activation group's dial code, then the Play List will activate whenever that number is sent via syslog.

It only happens during all call…..

If low audio, distortion, broken, intermittent or choppy audio only occurs during group or all call announcements (including voice announcements, bell tones or any audio sent to groups of speakers), but broadcasting to those same speakers individually results in clear, adequate sound, then power should be evaluated. Speaker power supplies, amplifiers or PoE switches may be failing or overloaded. Powering one or a few speakers during announcements does not tax the power source as much as powering all the speakers. To troubleshoot, either reduce speaker volumes or physically remove some of the speaker load from the PoE switch, power supply or amplifier and check to see if the clarity of the group/all call audio improves. Everything has a limit. Power supplies, amplifiers and PoE switches can only provide their rated amount of power. In the case of the latter, PoE switches often have a power budget. If this is the case, you may not be able to use all the available switch ports to provide full rated power.

Also refer to:

The importance of multicast in Valcom VoIP systems

Using Wireshark to Analyze VoIP RTP Audio

Wireshark may be used to analyze the audio sent to and from VoIP endpoints and gateways.

Here is a video demonstrating analyzation of SIP audio packets. and here is a video of analyzation of Valcom audio packets. Audio can also be exported to other programs, like Audacity, for analyzation.

Note that, analyzing UTP traffic requires that the PC running Wireshark be connected to a switch port that mirrors the switch port of the endpoint/gateway being evaluated.

Troubles Following a Power Failure

Troubles that follow power failures may be attributed to several issues. Overtaxed power supplies and amplifiers may not recover when presented with the higher startup load of the connected equipment. Check power supply output voltages to determine if voltage is at the rated level after a power failure. If not, reassess power supply loads and add power supplies or redistribute loads as necessary. For amplifiers, measure speaker loads and ensure that the do not exceed 80% of the amplifier's rated output.

Valcom VoIP devices can be set to acquire IP addresses from a DHCP server or may use static addresses. When programmed for DHCP, Valcom VoIP devices will revert to a default static address in the 192.168.6.X subnet if DHCP service is unavailable. This can cause trouble following a power failure. In such instances, the Valcom VoIP devices may not be routable to the DHCP server for an extended period, ten minutes or more has been observed. Therefore, they will revert to static addresses as described and will no longer function as desired until valid IP addresses are assigned. This is rarely an issue on robust networks that have alternate power backup for the DHCP server and all the switches and routers.

If this occurs, once the network is fully restored and stable, power cycle the Valcom endpoints - either manually by cycling power or, for PoE endpoints, by cycling the ports on the assigned switches ports. Valcom servers, will need to be power cycled by momentarily pressing the front panel power button with a straightened paper clip. Once the server shuts down, press the silent button to silence the power loss alarm and press the front panel power button with a straightened paper clip a second time to reinitialize the server. Repeat for each Valcom server in your system.

Assign static IP addresses or evaluate power backup plans to prevent this trouble in the future.

Reverberant Sound

Reverberation is a function of an area's acoustic characteristics (RT60). The ultimate solution is always acoustically treating the area with sound absorbent materials. Short of that some simple suggestions include:

  a) Use low mounted ceiling or suspended speakers spaced at the mounting height.
  b) When using high powered horns, use as few as possible, or in noisy areas, space them at <= twice the mounting height and angle them straight down.
  c) If personnel only occupy a small area, place a speaker or horn in that area so that they are in the direct field of sound.
  d) Direct all points of sound in the same direction.
  e) Angle horns directly down or directly up (if protected from weather).
  f) Turn speakers/horns down to the same minimum acceptable volume

The goal is to achieve minimally adequate and even sound pressure (volume) throughout an area. Any excess sound energy will not benefit the listener and only adds to the reverberation. Refer to our Frequently Asked Questions for more information.
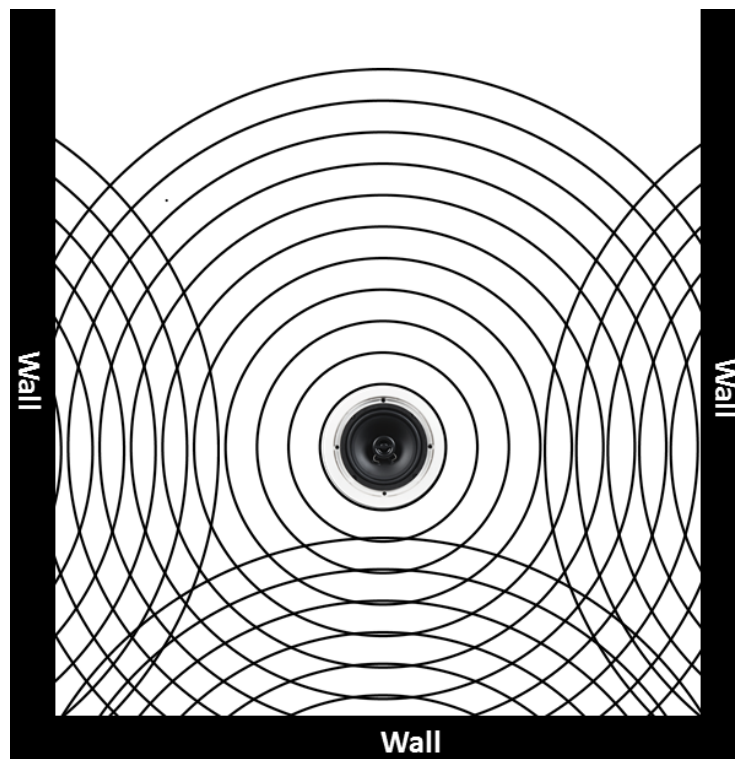
Delayed sound is often mistaken for reverberation. When delayed sound is noticeable, a listener will hear sound from a close sound point before they hear sound from a further sound point. Sometimes an area is plagued with both delayed sound *and* reverberation.

Delayed sound is a function of an individual's proximity to two or more sound points (typically horns) that are broadcasting the same audio. If the individual's distance from each sound point varies by ≈40 feet or less, then the listener will not perceive sound delay from the farther sound point (Haas Effect). Otherwise, any audible sound level from the further sound point may appear as an echo. Possible solutions:

a) If possible, place all sound points in a single location directed outward in a 360-degree dispersion pattern.
b) If using horns, angle them down and/or, horn placement permitting, turn the volume down so that the sound level at a point equidistant from all other sound points is no more than 3-dBspl above the ambient noise level.
c) If the listeners are always a fixed distance from the sound points, then delay lines may be used to delay the sound points closer to the listeners.

A

Horn A volume adjusted to reach Point 1

B
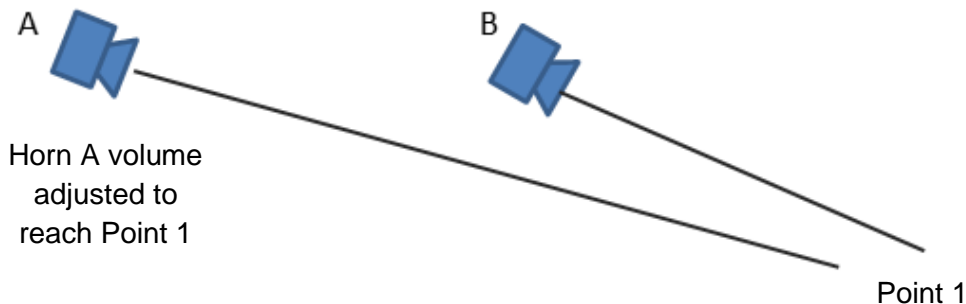
Point 1

Sound travels at approximately 1,125 feet per second. A listener at Point 1 will hear the sound from horn A after they hear the sound from horn B.

A

Horn A volume adjusted to reach Point 2

B

Point 2   Point 1
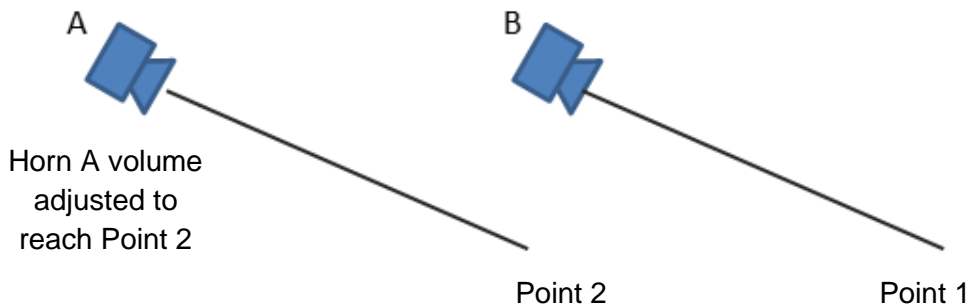
Angling horn A so that the direct field of sound is pointed below horn B takes the listener at Point 1 out of the direct field of horn A's sound and lessens the effect of the delayed sound.

Audio systems that transport audio digitally over a computer network have natural latency relative to the live broadcast. This is due to the time required to encode and decode the audio, this latency is only noticeable when the latent sound is audible from the page initiation point. In cases where it is audible away from the page initiation point, it is caused by network issues. Some network issues result in the multicast packets being sent twice. This can be diagnosed by a network analyzer such as Wireshark.
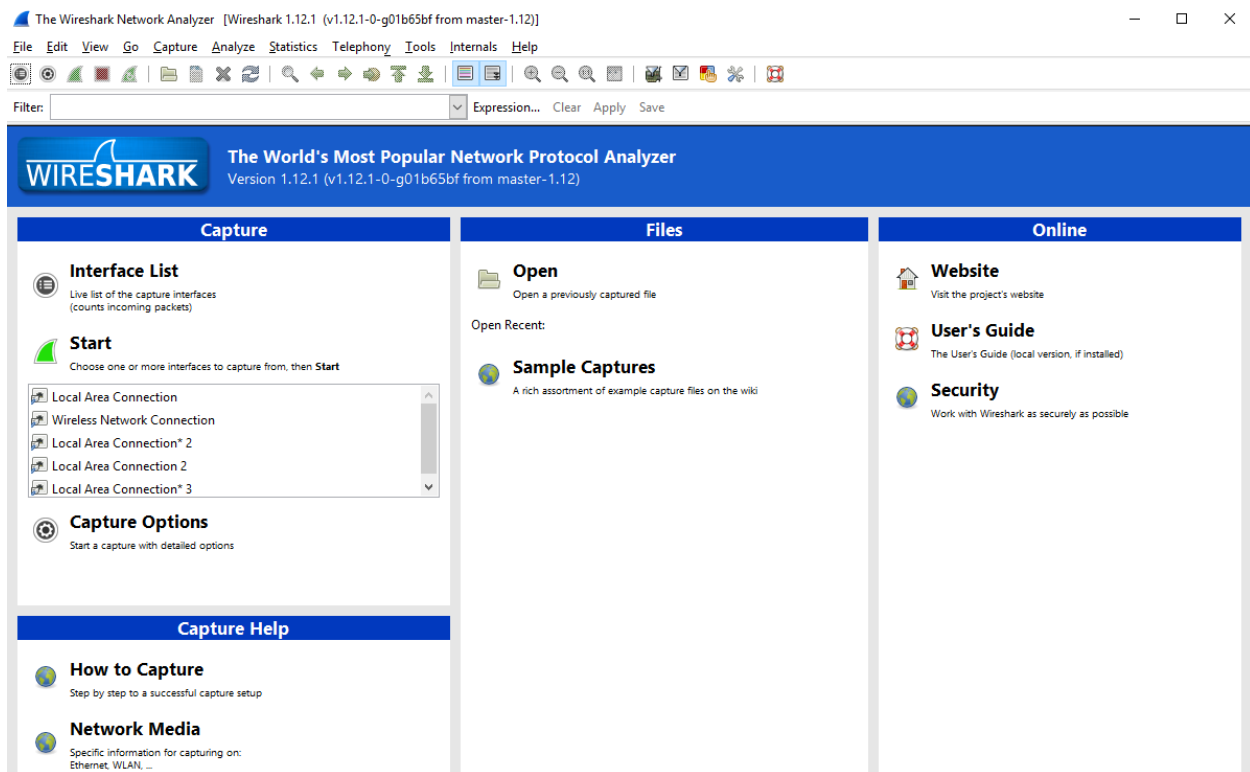
Acoustic Feedback (squeal)

When a live page is broadcast in the same area from which it is initiated, acoustic feedback can be an issue. Digital feedback eliminators and Application Servers record announcements and delay their broadcast until the initiating telephone or microphone is idle. Therefore, there can be no feedback. Valcom's Feedback Eliminators and Application Servers also allow page stacking so that multiple announcements can be recorded, queued and broadcast sequentially.

Other common, but less desirable remedies include turning the speaker volumes down near page initiation telephones/microphones or creating separate "all call" and other groups per affected area omitting the speakers in that area from the "all call" group.

Delayed broadcast feedback elimination techniques are only suitable for page initiation to one-way zones or groups.

Hybrid Echo (also see [Testing Talkback Switching](#))

Most public telephone system local loop wiring is done using two-wire connections - the same pair of wires carries voice signals in both directions. In the telephone company's central office or in an office PBX, a two-to-four wire conversion is done using a hybrid circuit. Hybrid circuits do not perform perfect impedance matches. The imperfection results in telephone users hearing their own voice reflected, or [echoed](#), back. Click [here](#) for more info.
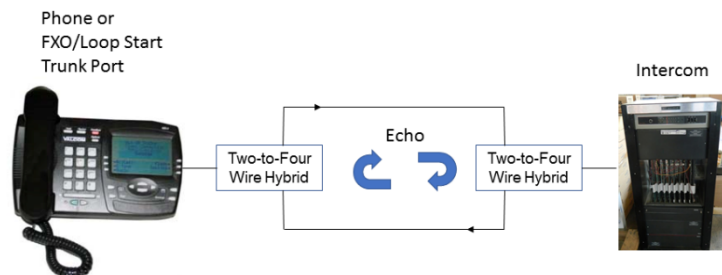
Impedance mismatch problems can also cause DTMF recognition issues.

This same phenomenon, although uncommon, can occur in talkback paging systems. Users at the telephone end or the speaker end of the conversation may hear their voices echo back. In our case, our intercom is the "central office" as we provide dial tone and talk battery. Both the phone system FXO/Loop Start trunk port, or the stand-alone telephone have two-to-four wire hybrid circuits that may be mismatched.

To remedy cases where users hear their own voice echo back:

(1)   To eliminate echo in talkback speakers, or breaking (choppiness), increase attack and/or decrease speaker-to-phone volume. Too much volume level from the speaker can cause the phone to act as if the phone user is speaking, thus giving direction control to the phone and causing breaking-up.

(2)   To prevent choppiness (when pausing between words), increase the release time value.

Setting the attack level too high or the release time too low can cause echo in the telephone receiver.

Some products, like Valcom's VERCA product, have an echo cancellation training mode built in to adjust for the originating phone's or port's impedance. If different types of phones and/or phone systems are used to access the same intercom, you will be able to train for one only.

Fortunately, there are commercially available products that can adjust the impedance of the second phone or phone system trunk port so that echo free communication can be enjoyed by all users. One such product is the [Echo Stopper](#) from http://www.sandman.com.

Uneven audio coverage

[Balancing audio](#) in systems that offer multiple volume settings (channels, group offsets, event offsets, etc.) does not have to be difficult. An important step is choosing a baseline around which all other volumes are adjusted. For example, all self-amplified speakers should all be installed with their integrated volume controls set for approximately ½ volume. Centrally amplified speakers should all be tapped equally.

Next, choose areas that are representative of most of the facility, find acceptable volume settings for one of these areas and mirror those settings to all similar areas. Note that on 45 Ohm Class Connection and MultiPath systems, all call is not adjustable at the system level. Therefore, talkback speakers should initially be set for "normal" or mid-range volume and all call must be used as your baseline volume.

Once you've established a baseline audio level, fine adjustments can be made on a per area basis as required.

Audio coverage by speakers and horns is a function of mounting height and the area's [ambient acoustic characteristics](#) and noise level. Refer to the following recommended [spacing chart](#). Note that, for horns, the chart is based upon an 18-foot mounting height.

Imperial Units of Measure

Speaker & Horn Placement Guide						
Mounting Height x 2 = Ceiling Speaker Placement	**Horns**		Quiet 50-65 dB	Moderate 65-80 dB	Noisy 80-90 dB	Very Noisy 90 dB+
		5 Watt	110' (12,000 ft²)	80' (6,400 ft²)	50' (2,500 ft²)	30' (900 ft²)
Wall Speaker Placement Spaced 20' Apart		15 Watt	-	-	75' (5,600 ft²)	45' (2,000 ft²)
		30 Watt	-	-	-	60' (3,600 ft²)
(1 per 600 ft²)		*Shows space between horns and coverage per horn*				

Metric Units of Measure

Metric Speaker & Horn Placement Guide						
Mounting Height x 2 = Ceiling Speaker Placement	**Horns**		Quiet 50-65 dB	Moderate 65-80 dB	Noisy 80-90 dB	Very Noisy 90 dB+
		5 Watt	34 m (1,115 m²)	25 m (595 m²)	15 m (232 m²)	9 m (84 m²)
Wall Speaker Placement Spaced 6m Apart (1 per 56 m²)		15 Watt	-	-	23 m (520 m²)	14 m (186 m²)
		30 Watt	-	-	-	18 m (334 m²)
		Shows space between horns and coverage per horn				

Phasing Troubles

It's a good practice to use the same polarity audio for all amplifiers and all speakers in an area. This is accomplished by wiring them all the same way:

>   a) If multiple amplifiers are serving the same area and share a common audio source then connect them all to the audio source output in exactly the same way.

>   b) Always connect the inputs of neighboring speakers (marked + on some speakers) with the same exact polarity.

Speaker cones push out and pull in as the applied signal voltage changes from positive (+) to negative (-). This movement pushes and pulls the air and causes vibrations that we interpret as sound.

If all speakers are in phase, then they are all pushing or pulling the air at the same time. Since the pressure waves are all going in the same direction, the sound pressures combine and reinforce each other.

If neighboring speakers are out of phase with each other, either because of their wiring, or because they are connected to out of phase amplifiers, then their sound pressures negate each other.

In Phase Audio   **Out of Phase Audio**

When working with old fashioned centrally amplified systems (100v, 70.7v or 25v), uneven sound coverage is often the result of improper design resulting in a significant differential in sound pressure from identically tapped speakers. For any given speaker wire run, identically tapped speakers closest to the amplifier are always louder than those further down the line. A professional audio design engineer will have taken these losses into account in the initial design so that all speakers will provide enough audio to acoustically overcome the ambient noise where they are installed. However, it's common for the function, and therefore the ambient noise level in different areas to change over time. Users attempt to compensate by adding more speakers and/or re-tapping existing speakers. This changes the loss of the line from the original design and affects the audio level on all other speakers on the line.

Note that Valcom's V-1095 70 Volt Expander works on 100v, 25v or 70.7v speaker lines to allow the addition of Valcom Self Amplified Speakers without affecting the line's balance.

25 Volt | 25 feet / 20 AWG Shielded | 5 Watt — Actual Wattage = 4.82 W | 100 feet | 5 Watt — Actual Wattage = 4.41 W | 100 feet | 5 Watt — Actual Wattage = 4.15 W | 100 feet | 5 Watt — Actual Wattage = 4.02 W — **19% Signal Loss**

25 Volt | 25 feet / 20 AWG Shielded | 5 Watt — Actual Wattage = 4.75 W | 100 feet | 5 Watt — Actual Wattage = 4.13 W | 100 feet | 5 Watt — Actual Wattage = 3.68 W | 100 feet | **15 Watt** — Actual Wattage = 10.07 W — **32% Signal Loss**
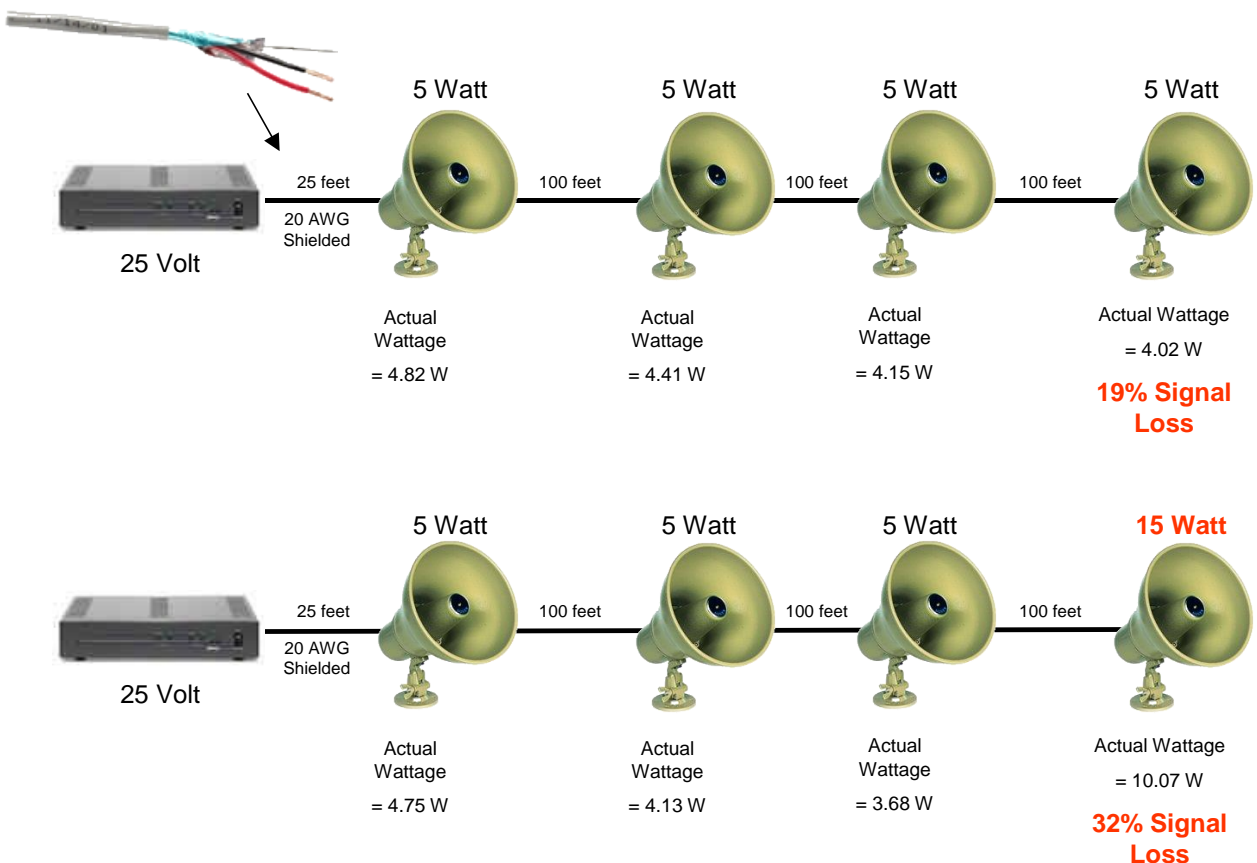
Simply changing the tap setting of a single speaker/horn, or adding more reduces the volume at all speakers!

Inadequate audio coverage

Inadequate audio coverage is a function of a design that requires:

a) More speakers or horns
b) Louder speakers or horns
c) Better placement of speakers or horns

Most manufacturers offer [guidelines](#) for sound point (horn/speaker) type and spacing vs. audio coverage for <u>typical</u> areas. However, these guidelines cannot take every variable into account. There is a whole engineering discipline, as well as modeling software, built around choosing audio sound points based upon an area's dimensions, the materials used in construction (every material has a coefficient of sound absorption), how those materials are finished, (unpainted concrete block walls absorb sound while painted concrete block walls do not), the expected content of the area (high sound blocking shelves, sound absorbent inventory, etc.) and anticipated ambient sound levels. In many areas, these variables change on a regular basis requiring dynamic audio adjustment via automatic volume controls. Because of the unique qualities of each area, a manufacturer's recommendations for typical areas must often be tailored.

Some basic knowledge of sound will help in correcting for inadequate coverage:

a) For voice announcements, the audio from a public address system should overcome the ambient noise level by at least 6-dBspl throughout any area (at average ear height).
b) Sound pressure (volume) from a sound point is typically greatest directly in front of the dispersion cone.
c) Doubling the distance away from a sound point will result in approximately 6-dBspl less sound pressure.
d) Doubling the power to a sound point will only yield an additional 3-dBspl of sound pressure.
e) A 3-dBspl difference in sound pressure is barely perceptible, a 10-dBspl difference in sound pressure is perceived as twice as loud.
f) Sound pressure also diminishes as listeners move away from the on-axis sound directly in front of the dispersion cone. The total angle over which the sound pressure level diminishes by 6-dBspl compared to the direct sound directly in front of the dispersion cone (on axis level and at some given frequency) is known as the dispersion angle.
g) When dispersion angles of adjacent sound points meet, in phase sound pressures will sum resulting in a 3-dBspl boost (sound will only be 3-dBspl down). This is the basis of the manufacturer's [spacing](#) charts for typical areas.

h) Equal sound pressure that is 180 degrees out of phase will null or cancel. This can occur when speakers are wired backwards from each other.

i) Manufacturer's [spacing](#) charts are based upon horns being mounted approximately 18 feet high.

No Voltage from Power Supply

If you determine that a system power supply is not providing output voltage, remove the wiring from the output, cycle power to the supply, and test again. If you measure voltage on the unterminated output of the power supply, then there is either excessive load on the supply, or there is a short circuit in the power cabling.

Phantom Calls

Phantom calls are call-ins from talkback speaker locations that occur without initiation. A common cause is when one of the conductors for the call switch wire pair is shorted with one of the conductors in the audio wire pair. Phantom calls often occur immediately after a general announcement (like all call). Troubleshooting involves temporarily removing the call switch wiring for the station in question and then trying to reproduce the phantom call. Call switch wiring with intermittent short circuits can also cause this phenomenon. Remember to always *individually* tape and store all unterminated conductors.

The author recalls a retrofit site where the integrator extended existing 3 conductor + shield cabling with UTP. In the classrooms, the shields were used as one side of the call switch. Poor wiring techniques in the existing wiring resulted in the shield periodically touching one side of the speaker wiring thus causing phantom call ins. The phantom call-ins were very sporadic and most likely caused by vibrations from air handling equipment.

Call Switch Troubles

If *simple* call switches fail to work, disconnect the call switch cable from the equipment and use your Ohmmeter to verify that the button produces a short circuit when pressed. This measurement should be made on the actual call switch and at the equipment end of the call switch cable. If you measure a short circuit at the button location, but not at the equipment end of the call switch cable, then assess the call switch cabling for an open circuit. If you measure a constant short circuit (even when the button is not pressed), then assess the call switch and it's cabling for a short circuit.

For a call switch to work, the equipment to which it is connected must provide some level of voltage on the call switch input. Disconnect the call switch wiring and use your voltmeter to verify that the equipment input does provide some level of voltage. If it does not, then verify all connections. Call tech support if the issue remains unresolved.

Some IP system call switches use resistors to indicate the auto destination (FXS channel or SIP identity) to be called. Use your Ohmmeter to verify the resistor values and the switch operation. Refer to the installation manual to verify connections.

Call switches are typically programmable to call a specific administrative port (analog intercoms) or FXS channel (IP systems). Verify programming. If pressing a call switch on an IP speaker results in an "all circuits are busy at this time" message, then the channel dial code in the auto destination field of the IP speaker (as seen in the VIP-102B IP Solutions Setup Tool) is not a valid auto destination or the gateway to which the IP speaker's call switch is auto destinated is no longer routable to the IP speaker.

For IP call switches auto destinated to unreachable SIP identities, there is no message.

Crosstalk

Crosstalk, hearing audio intended for other audio paths, is not common in Valcom systems due to the low audio transmission signals involved. When it occurs, there are some typical reasons:

a) Splitting UTP pairs. The intent of using UTP wiring with a differential audio input is to reject noise. Crosstalk is considered noise as it is undesired signal. Not maintaining the integrity of the twist in UTP renders it ineffective. The twist in UTP pairs should be maintained right up to the termination points.

b) Signal output wiring and signal input wiring routed in parallel. The electromagnetic field surrounding conductors carrying ac signals (audio) increases in strength following audio signal strength. This electromagnetic field induces crosstalk into surrounding conductors, including inputs to other circuits.

c) Improper grounding can cause crosstalk.

d) On rare occasion crosstalk can occur before the wiring is ever a factor. Not enough physical separation between the equipment that is the source of the undesired audio and the equipment that is the recipient of the undesired audio (crosstalk) may experience crosstalk. Audio transformers can both induce and receive electromagnetically coupled audio and noise.

e) Excessive signal in the source wire pairs can cause crosstalk. Turning source signals down in the cabling inducing the crosstalk and compensating at the load end (speaker, audio input) is often a remedy.

f) Low signal to noise ratio in the cables where crosstalk is observed. Increasing desired signal may make the undesired crosstalk signal less evident.

g) Unterminated audio outputs/speaker lines may contribute to crosstalk.

h) Excessive cable lengths and loops in cables. Keep cables as short as is practical and avoid loops.

Note that the audio level on the loaded audio source outputs, such as speaker outputs from intercoms or amplifiers, is typically much lower than on unloaded audio source outputs. Technicians troubleshooting crosstalk may be inclined to isolate cables by disconnecting speakers or speaker lines. The higher signal levels of these unloaded audio outputs can actually *cause* crosstalk.

When testing for crosstalk, the disconnected outputs should always be terminated with resistors sized to represent the actual load. Connect these load resistors as close to the audio output as is practical.
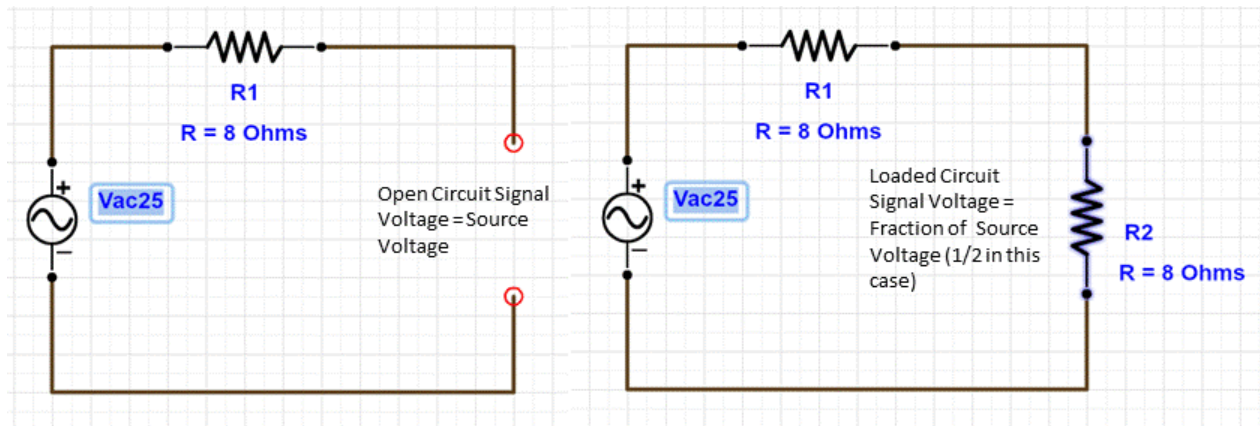
Troubleshooting crosstalk involves determining its source(s) and its destination(s). The author experienced a site where crosstalk occurred whenever certain combinations of zones were called as audio groups. While it appeared random at first glance, a pattern soon emerged. To troubleshoot, a new audio group was created by adding the speaker zones one at a time and testing for crosstalk after each addition. If crosstalk was present, then the last speaker added to the group was noted and removed from subsequent tests. In this case, 5 speaker zones were identified that caused crosstalk. Those 5 speakers were the only ones installed by a particular technician. He thought that the speakers needed 25-volt transformers, so he added them. He had left the unused speaker taps uninsulated and all touching the ceiling tile grid thus creating the crosstalk path.

Cannot connect to network-based equipment

First, Network connection and endpoint scanning troubles may occur if the scanning PC and/or the endpoints are plugged into network switch ports that are using MAC address filtering or some other form of MAC address security that is denying access to the Valcom endpoints or the scanning PC.

Next, refer to the *Reboot Trouble or Reboot Required Troubles* section of this manual.

If the problem remains unresolved, then cycle power or properly reboot the device to reestablish network connectivity. Connecting to network-based equipment requires:

a) That your PC be routable to the network-based equipment. If this appears to be correct:
   1) Using your non-configured multiport PoE switch, assign an IP address to your PC that is on the same subnet as the equipment and connect both your PC and the equipment to your test switch. If you can connect using this method, but not through the network, then the problem lies in the network configuration. If you cannot access using this method, then try pinging the network-based equipment to verify its IP address.
   2) Make certain that the subnet mask and gateway are valid for the IP addresses in use.

b) That your PC is utilizing the correct NIC
   1) Disable all unused network interface cards, such as wireless cards
      i. On some PCs turning off the Wi-Fi from the system tray doesn't always turn it off on the backend, it may have to be disabled in the Services menu.

c) That your PC is not blocking required ports
   1) Temporarily disable your firewalls and any port blocking (antivirus/PC protection) software.

d) For non-browser-based access, that you are using the right PC software tool.
   1) Class Connection ES and Multipath software tools are unique. Using the wrong one will result in connection failures.

e) That there are no IP address conflicts.
   1) Disconnect the network-based equipment and ping its address on the network. If you receive a response, then there is an IP address conflict. The author has experienced situations where an IP address assigned to Valcom equipment was also assigned to a computer that was not always turned on. In this case, connectivity and functional issues only occurred when the offending PC was turned on. Assume nothing.

f) That all infrastructure is properly terminated. Poorly crimped RJ45s have often been the root cause network connectivity troubles.

g) That your PC be connected to an actual switch port and not through an Ethernet port on an IP telephone or the like.

h) That a DHCP server is available, if required.

i) That the endpoint's network capabilities are compatible with the network switch port. For example, 10 Mbps ½ duplex may not work on 100/1000 Mbps full duplex switch ports.

There are network requirements available for all Valcom VoIP devices. To ensure proper operation, it's important to adhere to these requirements.

Endpoints won't communicate with each other

Network communication troubles may occur if the scanning PC and/or the endpoints are plugged into network switch ports that are using MAC address filtering or some other form of MAC address security that is denying access to the Valcom endpoints or the scanning PC.

Intercommunication between network-based endpoints requires that the devices be routable to each other and that all necessary ports and protocols are open/enabled. If the network settings appear to be correct, then connect the endpoints to your non-configured multiport PoE switch, If the endpoints communicate using this method, but not through the network, then the problem lies in the network configuration. If you cannot access using this method, then verify all endpoint network credentials and make certain that the subnet mask and gateway are valid for the IP addresses in use.

There are network requirements available for all Valcom VoIP devices. To ensure proper operation, it is important to adhere to these requirements.

Managed Switch Port Works Selectively

If 2 or more endpoints are sharing a single port on a managed network switch (perhaps utilizing a second unmanaged switch) and only one endpoint can successfully communicate on the network, the managed switch's port may be using MAC address filtering or some other form of MAC address security and denying access. The managed switch may lock on to the first MAC address it encounters and not allow access to any other endpoint.

Some reasons why endpoint network communication fails

➢ The endpoint has invalid/incorrect network credentials

  o The endpoint has failed to obtain network credentials via DHCP
  o Incorrect network credentials were manually assigned

➢ The port that the endpoint is connected to is blocking traffic because the endpoint is not whitelisted (MAC address filtering).

➢ The port that the endpoint is connected to is blocking traffic because the endpoint has unsuccessfully tried to connect too many times (denial of service protection).

➢ The endpoint, its patch cable or switchport is defective or not connected

➢ The endpoint is not powered or is underpowered (i.e., PoE+ endpoint on PoE)

➢ Mismatch of network speed (i.e., endpoint is 10 Mbps ½ duplex switchport is 100 Mbps full duplex)

➢ The endpoint is trying to connect to an endpoint on a different subnet and does not have a valid gateway defined in its network credentials.

➢ There is no proper route defined in the network(s) to allow connection to supporting endpoints, originating or destination endpoints. Therefore, the endpoint is not routable to the VLAN or subnet used by endpoints to which it is trying to communicate.

➢ The endpoint cannot resolve the domain names of destination endpoints because no valid DNS servers are available.

➢ Required protocols are not, or are no longer, available to allow connection to supporting endpoints, originating or destination endpoints. This could be related to PIM or Spanning Tree.

➢ The endpoint it is trying to communicate with is not reachable.

➢ Network congestion

➢ If the communication issue is only at boot up, such as failure to retrieve NTP, or register with another system at boot, then spanning tree may be in use and the port is not ready. Defining the port as an Edge Port or PortFast may help.

➢ Access is restricted/limited because the endpoint has a static address and network security parameters only allow dynamically assigned addresses.

➢ IP address conflict. IP address conflicts can hamper network efficiency and temporarily disable operations until they are resolved.

➢ Some network DHCP servers will restrict access to the Internet or other resources for endpoints that are set for static addresses.

USB WAV File Upload Troubles

We have seen instances where certain PCs have trouble transferring WAV files via the USB connection on certain products. Ensuring that you have the latest USB driver installed and using a simple USB hub between the PC's USB connection and the system USB I/O has been known to alleviate this trouble.

Other PC Connection Troubles

Every PC that has been in service for some period of time is unique. Occasionally the programs that have been installed on a PC or PC settings may cause connection troubles. If troubles persist after disabling all unused network interface cards (wireless and other) and firewalls/protection software, the fastest course of action is to try a different PC. Note that on some PCs turning off the Wi-Fi from the system tray doesn't always turn it off on the backend, it may have to be disabled in the Services menu.

VIP-102B Scanning Troubles

Valcom endpoints and gateways are discovered by the VIP-102B in several ways:

1) By any IP addresses defined in the network setup
2) By the endpoints responding to a broadcast roll call request
3) By the endpoints or gateways responding to the tool's multicast roll call request
4) By the tool receiving a multicast beacon from the endpoint or gateway
5) By the tool receiving a broadcast beacon from the endpoint or gateway

If IP endpoints and gateways are not discovered by the VIP-102B IP Solutions Setup Tool, **then make certain that:**

1) You are using the latest version of the VIP-102B
2) You are not inadvertently using the "Valcom IP Tool"
3) The endpoints are powered
4) Multicast is fully implemented
5) The network requirements have been met
6) All switch ports are on a dedicated port based VLAN, or are part of a VLAN dedicated to both Valcom VoIP endpoints and VoIP telephony traffic
7) The PC's NIC currently utilized by the VIP-102B is connected to the VLAN
8) The PC is not connected through an IP phone auxiliary Ethernet port
9) All unused network interface cards on the PC are disabled
10) All port blocking protection software on the PC is disabled
11) Connectors/cables/switch ports are not defective
12) RJ45s are properly terminated

13) The tool Network Settings (File menu) match the endpoint or gateway network settings. Endpoint or gateway network settings are typically set to default values, however, can be changed on the System/Device Network Settings Menu and on each endpoint or gateway Network Tab

14) The IP endpoints and gateways are not listed in the VIP-102B's Ignore List

15) Any virtual machines using VMWare are disabled on the PC

16) If using spanning tree, that the port is defined as an Edge Port[1]

17) The scanning PC and/or the endpoints are not plugged into network switch ports that are using MAC address filtering or some other form of MAC address security and are, therefore, denying access to the Valcom endpoints or the scanning PC.

18) The PC's Ethernet connection does not utilize a USB to Ethernet Adapter (dongle)

19) The endpoints have successfully retrieved an IP address from DHCP or have obtained a static IP address.

Red "error status" ICONs indicate a routing issue between the computer hosting the VIP-102B and the endpoint(s). They may also be the result of IP address conflicts, invalid DNS entries having been assigned to the endpoint(s) and/or a time out issue requiring adjustment in the VIP-102B to facilitate waiting longer for endpoints to respond.

[1]The spanning tree algorithm is used in switch-based networks to determine the best path for traffic to move between two network devices.

Spanning tree algorithm prevents loops and promotes efficient data delivery by discovering all possible paths between two ports on a network and ensures that only one is active at a time.

If the primary path is not available, then spanning tree protocol directs traffic to alternate paths if available.

Spanning tree protocol utilizes a primary decision-making switch called the root switch. All switches in a network utilizing spanning tree protocol branch from the root switch and all network paths are traceable back to the root switch. The root switch is necessary as it serves as a reference point for the best route calculations.

Data messages exchanged between switches, known as bridge protocol data units (BPDUs), gather information about other switches in the same network.

Each port on a switch using spanning tree protocol may be in any of five states – blocking, listening, learning, forwarding or disabled.

With spanning tree enabled, every switch port in the network starts in the *blocking* state, then moves to the *listening* and *learning* states. If a potential loop is detected, the port will be set to the *blocking* state. If no loop is detected, the port will be set to the *forwarding* state. The ports then sustain the forwarding or blocking state unless changes are made to the network.

The bridge protocol data unit exchange is used to determine switch in the network will serve as the root switch.

A switch always enters the blocking state following switch initialization and will not send or receive any data traffic across the network segment. A port in the blocking state will however listen to BPDU messages. This could create issues with the VIP-102B device discovery.

If you wish to force a port into forwarding state when connected, bypassing the blocking, listening and learning states, then program the port in Edge Port/PortFast Mode. (note: Unlike PortFast, a Cisco term, an edge port that receives a BPDU immediately loses edge port status and becomes a normal spanning tree port. There are no additional key differences between these features.) A port is considered as an edge port when it is directly connected to the user terminal or server, instead of any other switches or shared network segments. The edge port will not cause a loop upon network topology changes.

Incorrect System Time

Most Valcom systems that require accurate time have some provision for obtaining that time and automatically adjusting it for time zone and Daylight-Saving Time (DST). Sometimes fixing incorrect time on a Valcom system is as simple as entering the correct time server address, time zone or DST setting. Sometimes, is it related to the network settings.

Class Connection ES (prior to rev 5.08) and MultiPath systems may be directed to a Daytime Protocol Server. They correct once a day at a time defined by the user.
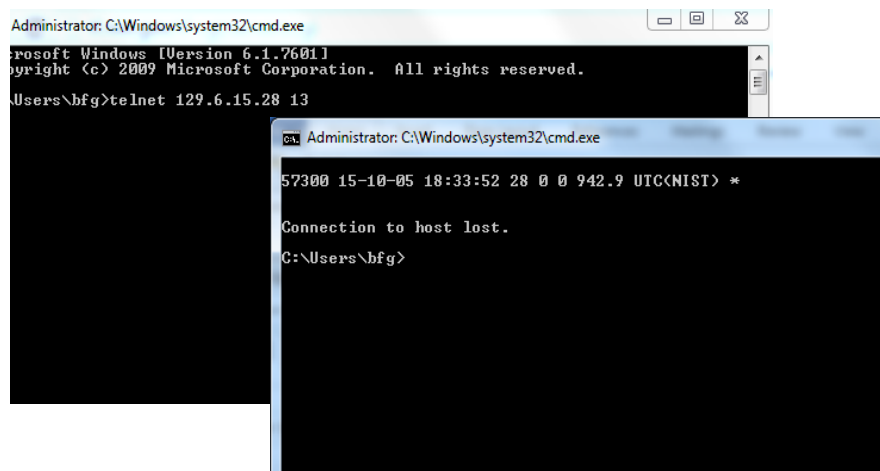
Valcom IP servers and equipment use Network Time Protocol (NTP) version 4, they correct on boot up and periodically throughout the day. Some NTP servers only provide NTP version 3. Contact us if the only NTP server available on site provides NTP v3.

Although both services may be available from a single time server, these are 2 different protocols and are not interchangeable. NTP uses port 123 and daytime protocol uses port 13.

Regardless of whether your time server of choice is part of the local network, or located on the Internet, the network port connected to the Valcom equipment must be properly routed to that time server to successfully obtain time. If resolving a domain name (like us.pool.ntp.org), then valid DNS entries will be required for the endpoint attempting to obtain time.

Time servers may be tested by plugging a PC into the switch port that was being used by the Valcom endpoint, setting the PC's network credentials (IP address, gateway, subnet mask) to match the Valcom endpoint and disabling all PC network ports except for the one connected to the switch. Refer to the following examples:

If testing a Daytime Protocol server, telnet to the server on port 13 and it should return a time string:

Windows PCs may be [configured to serve daytime protocol](#). A PC serving daytime protocol to Valcom systems must use a static IP address and must be operating and online when served devices request time. Note that you may encounter an "incorrect format" error, however, the time synch will work.

For an NTP server, first, [ping](#) the server's address to verify that it replies.

For some later versions of Windows (tested on Win 7 and 10*), if the NTP server is available, the command prompt: **w32tm /stripchart /computer:<IP of the NTP Server that you are using> /dataonly /samples:5** will return 5 time offsets between the computer and NTP time returned. In the author's testing, the time returned was automatically adjusted for the PC's time zone. There is also an NTP test program located [here](#).

Valcom servers may not get correct time immediately upon initial sync, for immediate correction the server's time must have a > 3-minute delta from NTP time when the NTP server is polled. In practice, the time will match and track the NTP time within a few hours.

If the Valcom server is being used to serve NTP, it will serve manually adjusted time immediately, however, NTP acquired time will not be served for 30 minutes.

As of this writing, us.pool.ntp.org (67.18.187.111) is a valid Internet based NTP server.

*Windows 10 required the higher privilege Command Prompt (Admin). Press the Windows key and x to access the menu offering command prompt access.

If you need a test NTP source, then you can install [NetTime](#).

NetTime is a Simple Network Time Protocol (SNTP) client for Windows 95, 98, Me, NT, 2000, XP, Vista, 7, 8, 10 and Server 2003/2008/2012/2016. (32 bit and 64-bit operating systems are both supported). There is an option on the Settings menu "Allow other computers to sync to this computer" that allows the PC running NetTime to act as an NTP server.

> REMEMBER - Proper time server testing requires that the PC used for the test be plugged into the same switch port that was being used by the Valcom endpoint, setting the PC's network credentials (IP address, gateway, subnet mask) to match the Valcom endpoint and disabling all PC network ports except for the one connected to the switch.

PoE Ports Shutting down

If PoE switch ports shut down or reboot, the powered endpoint may be drawing excessive current. For Valcom VoIP devices that source audio inputs (on amplifiers or other systems), this is sometimes a function of a ground loop between the audio output of the endpoint and the audio input of the receiving device. In this case, the ground loop should be isolated. Mismatches in ground potential, or ground loops, are easily remedied by routing the line level audio through isolation transformers. Valcom VMT-2 1:1 isolation transformers have solved many ground loop problems and should be standard issue to all audio technicians.

For Valcom VoIP devices that source audio to analog speakers, like 8 Port Gateways, excessive PoE current draw may be a function of excessive speaker load, short circuits or ground loops in the speaker cabling. Always pre-check speaker lines, new or existing, for transient voltage, short circuits to ground and proper impedance before interconnecting to the intercom equipment. Use a good quality **Impedance Meter** to measure impedance, not an Ohm meter.

Valcom VoIP audio devices draw very little power at idle. However, when they begin to broadcast audio the current required from the PoE port increases significantly. Some PoE switches have default settings that detect sudden increases in current draw and shut the port down as a means of protection against defective endpoints. If PoE switch ports only shut down following an attempted page announcement, then PoE power management features may need to be set to static or high priority. This pre-allocates power to the endpoint, even when power requirement is at a minimum. This guarantees that when the endpoint requires more power, it will be available.

Additionally, PoE+ devices use LLDP-MED protocol to advertise their extended power preference. Some LAN switches that cannot provide the extended PoE+ power may shut down or reboot the network port. On some PoE switches, the default configuration is for PoE requests to be ignored if detected through LLDP or LLDP-MED.

Some PoE switches can provide PoE load information and PoE current, voltage and power testers are also commercially available.

Be aware that PoE switches have [power budgets](#) that must not be exceeded. Full power may not be available from all ports simultaneously.

We've also experiences PoE port shut down issues that were related to PoE switch firmware.

In one scenario, 8-port gateways were going into a constant reboot state where both Ethernet LEDs flash in sequence.

Not all 8-port gateways on a switch experienced the trouble. When switching ports with a working gateway, the problem followed the gateway. The Cisco WS-C2960X-48FPD-L switch logs did not show any data or PoE errors. The 8-port gateways functioned properly when moved to an unmanaged PoE test switch. Upgrading the Cisco WS-C2960X-48FPD-L switch firmware to version 15.2(4)E6 solved the trouble.

Radio Frequency Interference

Radio frequency interference (RFI) rarely occurs in Valcom Systems. The author has never experienced RFI in an IP based system and only occasionally in analog systems. Steps for troubleshooting RFI vary from case to case.

If your equipment is reacting to nearby amateur radio or CB transmitters, then you will have interference during the broadcast half of the conversation. If this is the case, survey the area surrounding your facility for an antenna mounted on a nearby house or car. Advise the Federal Communications Commission of your findings.

Some common steps in troubleshooting include:

a) Assuring that system power supplies and amplifiers, basically anything with an ac electrical cord, is connected to a direct Earth ground. Valcom power supplies outputs are not referenced to Earth ground, therefore a strap from the supply common outputs (the + output on -24vdc supplies) must be strapped to Earth ground. The 3rd prong in electrical outlets may or may not provide an adequate ground dependent up how a building ac power is wired. Bonding to a known good ground, such as a ground rod with as short as is practical 18 AWG+ gauge stranded copper wire is your best bet.

b) Connect all system ac power inputs through suitably sized [Tripp Lite](#) Isobar RF filtering surge suppressors. Follow Tripp Lite's instructions.

c) For shielded cable, ground the shields on one end only – the end closest to your originating equipment.

d) Earth ground the equipment end of all unused conductors in cables.

e) Terminate all unused audio inputs with a short piece of wire. Before doing so, verify that the inputs are not connected to any other circuit and have no measurable voltage across their connection points.

f) Make certain that equipment racks and metal enclosures are Earth grounded to the manufacturer's requirements.

g) Maintain the integrity of UTP wiring right up to its termination points. Use VMT-2 1:1 transformers on both ends of UTP cables to promote common mode rejection.

h) Install ferrite beads on audio cables connecting equipment. Ferrite beads work by increasing the serial inductance of the wire to effectively filter out high frequency RFI. There are many references to using ferrite beads in RFI elimination on the Internet.

i) Keep wire runs as short as possible.

j) Use good quality shielded cable for all microphone connections.

k) If known, contact the owner of the RFI source or the Federal Communications Commission for assistance. Unlicensed radio stations operating at higher than legal power have been known to cause RFI issues in the past.

l) If RFI is affecting self-amplified one-way speakers that are <u>only</u> connected to dc voltage, then with the speaker/horn Tip and Ring disconnected from the source, measure the dc voltage from:

a. Tip to GND
b. Tip to Earth GND
c. Tip to -24VDC or -48VDC
d. Ring to GND
e. Ring to Earth GND
f. Ring to -24VDC or -48VDC

All should measure zero. Next measure the resistance from:

a. Tip to GND
b. Tip to Earth GND
c. Tip to -24VDC or -48VDC
d. Ring to GND
e. Ring to Earth GND
f. Ring to -24VDC or -48VDC

All should measure infinity (open circuit).

If you measure otherwise, contact tech support.

If RFI occurs when certain audio inputs are connected, then low pass filters may be used as close as possible to the audio inputs of the system receiving the RFI. The filter shown below will roll off any frequencies above 23 KHz. The ground shown must be a direct Earth ground.
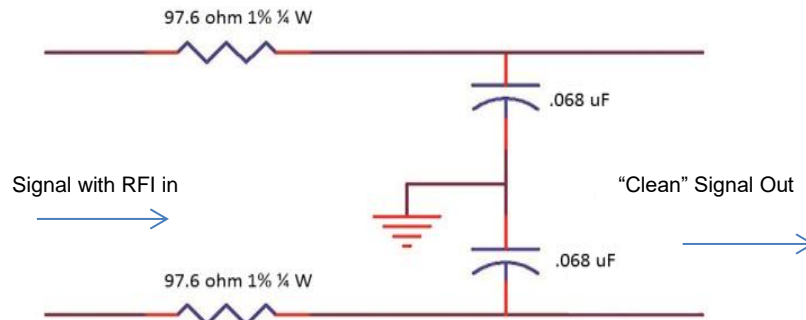
To eliminate the power supply as a contributor to the RFI, try powering the speaker from batteries instead of the DC power supply. 9-volt batteries are easily connected in series, each adding 9 volts to the total voltage. Valcom one-watt interior speakers may be temporarily powered by a couple of 9-volt batteries placed in series (18vdc).

Strong radio frequency interference (AKA Electromagnetic Interference) trouble can manifest itself in non-audible ways as well.

All Valcom products are designed to minimize the unwanted effects of RFI/EMI; however, in extreme situations (i.e. sites located next door to radio antennas) no level of design preparedness will be sufficient.

Any electronic equipment installed near strong RFI/EMI sources could experience ill effects including malfunctions at the component level.

Significant RFI/EMI sources include:

        a.   Analog television station transmitters
        b.   AM/FM radio transmitters
        c.   Solar Magnetic Storms
        d.   Lightning
        e.   Radar systems
        f.   Electrostatic Discharge
        g.   High voltage power lines
        h.   Arc welders (MIG and TIG welders as well)

On some occasions *removing* all Earth ground references from system power supplies, Earth grounding all equipment racks/enclosures, unused conductors in system cables or shielding may offer relief from RFI/EMI troubles.

Fortunately, due to Valcom's preventative design measures, RFI/EMI troubles are very rare and only occur in a very, very small percentage of installations.

DHCP Endpoints Reverting to Static Addresses

Valcom VoIP devices can be set to acquire IP addresses from a DHCP server or may use static addresses. When programmed for DHCP, Valcom VoIP devices will revert to a default static address in the 192.168.6.X subnet if DHCP service is unavailable. This can cause trouble following a power failure. In such instances, the Valcom VoIP devices may not be routable to the DHCP server for an extended period, ten minutes or more has been observed. Therefore, they will revert to static addresses as described and will no longer function as desired until valid IP addresses are assigned. This is rarely an issue on robust networks that have alternate power backup for the DHCP server and all the switches and routers.

Systems Randomly Stop Functioning

Valcom system software programming tools are always backwards compatible. Therefore, the latest versions of these tools should always be used. If you program a new system with an older programming tool, it may cause the system to randomly stop functioning. For example, a customer attended a training class in 2007. In the class, he received a copy of one of our programming tools. He never updated the programming tool and continued to use it for systems he installed 7 years later. In those 7 years we made several significant feature enhancements. The programming tool from 2007 did not support the enhancements and the new systems he installed would periodically stop working due to the incompatibility.

Other reasons that systems may periodically stop working?

a) Is the system or part of the system on a switched ac circuit? Is someone periodically turning off the switch?
b) Is someone occasionally unplugging the equipment to "borrow" the outlet?
c) Changes to the facility network?
d) IP devices falling out of the multicast group? (See Testing Multicast)

Also refer to:

The importance of multicast in Valcom VoIP systems

Cannot Access Valcom System

See Lineman's handset, VoIP Configuration Guides or SIP Registration errors.

It may be that you can access the Valcom system, but it immediately disconnects. Refer to the *Disconnect Troubles* section of this document.

If your Valcom System features the ability to call from speaker locations via call switches, and the ring cadence produced by the Valcom system is unusual (i.e. fast ring for emergency calls) then some phone systems will react by shutting down the port connected to the Valcom equipment until it is momentarily removed from the phone system or power cycled.

Reinitiating a system by cycling power will not provide information about the trouble, however, may clear the trouble. Having to reinitiate systems or individual Valcom VoIP devices on a regular basis is abnormal. If this becomes a common practice, then contact Valcom Tech Support for further assistance.

SIP (Session Initiation Protocol) Troubles

Valcom gateways and endpoints are configured for SIP access via the VIP-102B IP Solutions Setup Tool.

Most Valcom IP Speakers and Gateways are SIP accessible. To troubleshoot SIP, it is important to have a basic understanding of how it works. This document is a PDF that describes, in general terms, the flow of SIP between phones and Valcom endpoints/gateways programmed as SIP stations.

Almost all SIP troubles are related to network configuration, setup of the telephone system's SIP server, which varies by manufacturer, SIP registration expiration or invalid information being programmed into the Valcom endpoint/gateway.

Note that features and programming options vary by SIP provider. The following are general statements and may or may not apply to your specific application. Check with your SIP provider (internal or external) for system specific options.

Valcom endpoints/gateways utilizing SIP will register with the SIP server and suggest a registration expiration time of 3600 seconds. The SIP server may negotiate a different time, which the Valcom endpoints/gateways will honor. The Valcom endpoints/gateways will send re-registration at approximately 60% of the agreed upon registration expiration time.

When necessary, the default registration expiration time for Valcom endpoints/gateways can be changed to a higher or lower value. Lower values are particularly useful for hosted telephone systems. Using a value as low as 60 seconds may be necessary to keep firewall ports opened.

If dialing a SIP identity results in a slow busy, then the SIP endpoint being called is actually in use and cannot accept another call at the moment (like calling a busy phone line).

A fast busy can mean that the Valcom endpoint/gateway could not be accessed by the Telephone SIP Server. It is quite possible for a Valcom VoIP device to be registered with the SIP server and still return fast busy or a 404 error when called. This is indicative of improper routing from the Telephone SIP Server to the Valcom endpoint/gateway.

If applicable, Valcom endpoints require that the phone switch is set to send DTMF as "rfc2833" type or "rtp-event". Failure to do so will cause all, or some, dial codes to fail.

Additionally, the Telephone SIP server has settings to allow or disallow certain SIP phones from calling certain SIP endpoints.

If a SIP endpoint repeatedly attempts to register to a phone switch, such as what might happen when trying to figure out the correct credentials, the phone switch may blacklist the endpoint for some period of time. The period of time is dependent on the phone switch involved.

Some phone systems will not register SIP endpoints with multiple channels, such as multiport gateways.

The VIP Utility Tool may be used to collect the console output of a Valcom endpoint to a text file for further analysis. This text file will indicate the progress of SIP connections. The device

should be rebooted when the console session is ended. The default device password is "moonbase". Note that the text file may yield a more accurate rendering of results than the on-screen console output.

If a SIP endpoint *accessed by a hosted phone system* requires a reboot periodically to start working again, a firewall may be closing the connection during periods of inactivity. Adjusting the SIP Keepalive setting to a low value, perhaps 120 seconds may prevent this from occurring. Also, some Valcom endpoints have an Options setting that can be turned on to send Options to the SIP server. Setting this to a reasonably short value, perhaps 60 seconds may also remedy the firewall closing. This is generally not needed if the PBX is on premise or in a company's private network.

A tutorial for troubleshooting SIP with Wireshark may be found here.

Also see VoIP Configuration Guides, SIP Registration errors or Using Wireshark to Analyze VoIP RTP Audio

Caller ID/SIP Troubles

For providing a SIP interface to analog Valcom systems, FXO Gateway channels may be directly accessed via Session Initiation Protocol (SIP). They can also route incoming, ringing FXS connections (such as those initiated by talkback speaker call switches) to SIP phones. Once answered, the audio connection is a standard full duplex 2-way phone call connection.

Trusting caller ID from a SIP enabled device (when in Station mode) is a function of the telephone system's SIP server. In SIP Trunk mode, caller ID is usually trusted by default.

Analog Phone Caller ID Troubles

Caller ID information (CLID) is transmitted between the first and second burst of ring voltage. You should be able to hear the data with a lineman's handset connected the telephone's input (tip/ring) pair. The lineman's handset must be in monitor mode.

In addition, if phones do not display caller ID information, they may not be caller ID compatible or may use a different form of Caller ID. Valcom uses the telephone industry's Bellcore standard.

If phones only show the phone number, but not the description of the caller, then they may not be compatible with *enhanced* Caller ID. With enhanced CLID there are two lines that are transmitted in the CLID packet, however, if the receiving device is not "compatible" with the enhanced format, only one line will display. Test the tip/ring pair with a known working caller ID telephone.

SIP Trunk Mode Troubles

SIP Trunks do not generally require Valcom gateways to register. Routing for the extension(s) associated with the Valcom gateway need to be configured in the phone system, so that calls to your defined extension number(s) can be properly routed.

For Valcom gateways with an actual SIP Trunk Mode, like SIP Paging Gateways, if registration *is not* required by the phone system then you only need to configure the "Extensions" portion of the SIP trunk form. The SIP Trunk mode form may be found in the VIP-102B IP Solutions Setup Tool. Refer to the following figure:

For SIP Trunk Mode

Most phone systems only require the Extensions section to be populated.

A route must be built in the phone systems for each extension defined.

If registration *is* required, then populate all required fields, which vary by phone system. The registration of the Valcom gateway represents the SIP trunk. The SIP trunk registration can then be associated with multiple dial-peers (extensions) for routing outbound calls to Valcom Groups.

Note that any SIP endpoint can be configured as a SIP Trunk (even if it does not have a SIP trunk option). Typically, you would *not* check the "Register" option on the Valcom gateway's SIP form as registration is not usually required for SIP trunks. Registration, by properly completing the SIP form of the Valcom Gateway, is more likely required if the Valcom gateway is initiating audio (talkback).

A route must be built in the phone systems for each extension defined for calling to the Valcom endpoint/gateway as well as an inbound route coming from a Valcom endpoint/gateway that is talkback, to a specific extension or hunt group.

SIP audio is sent via Real Time Protocol (RTP). The port used is auto negotiated during setup via Session Description Protocol (SDP). If the negotiated RTP port is blocked by a firewall, then no audio will be received. Note that all SIP communication is UDP.

Busy Signal When Dialing an Intercom Station

When accessing through a Loop Start C.O. Line/Trunk port or POTs telephone, a busy may indicate that the station is truly busy. If dialing a code that begins with # through a Loop Start C.O. Line/Trunk port returns busy after the # is dialed, try substituting the digit zero for #.

Also refer to:

The importance of multicast in Valcom VoIP systems

Cannot Access Specific Features/Functions

For browser accessed Valcom VoIP devices, be aware that all browsers are not created equally. As of this writing, Internet Explorer does not support the microphone functionality of the Application Servers and Chrome does not support streaming audio. Firefox version 36 supports all functions.

Features and functions are associated with various levels of firmware revision. If an expected product feature is not available, or is not working as expected, then contact technical support to inquire about a firmware upgrade.

Browser Cache Overflow

When working with browser-based systems, your browser of choice will typically be caching (saving) pages. If browser cache gets too large, it can slow down the browser and cause other issues.

If the cache is being overloaded, run the browser in private or incognito mode to keep the browser cache as empty as possible.

Scheduled Bell Tones/Audio Not Working

Systems that feature the ability to broadcast scheduled audio rely upon:

a) The accuracy of the <u>system's time and date</u>
b) The correct programming of the destination audio group or groups. Are the recipient speaker zones/stations in the group(s) that is/are scheduled to receive the audio?
c) That automatic schedule operation is enabled (method varies by system)
d) That the schedules that have been created are using the correct time format (typically a 24-hour time format)
e) That the scheduled events (audio with a destination) have a very high priority (just below or at the highest emergency broadcast priority).
f) That the schedules have been assigned to operate on the correct days or dates (usually via a system calendar – refer to the system manuals)
g) That a manual voice announcement to the group destined to receive the scheduled audio is successful (does the group work at all?)

Reboot Troubles or Reboot Required Troubles

Refer to the *Intermittent Troubles* section of this document. If the trouble is specific to Valcom IP speakers and gateways not receiving one-way audio while channel dial codes still operate properly, or if a **Reboot is required to successfully call a channel** then the trouble may be related to network configuration.

If a Valcom IP speaker or gateway occasionally requires a reboot (power cycle):

1) to receive one-way audio (group announcements or audio from an Application Server for example)
2) to remedy channel dial codes fails typically resulting in a fast busy (*on certain devices*)

then it may not be receiving multicast. Refer to:

The importance of multicast in Valcom VoIP systems

Reboot required to obtain a new dynamic IP address

If a Valcom IP Gateway, Speaker, Server or other endpoint with an IP address obtained through DHCP periodically requires a reboot, but always has LEDs illuminated (indicating that it is still powered), then it may have reverted to a static address. Refer to the DHCP *Endpoints Reverting to Static Addresses* section of this document.

Troubleshooting Tip – Static IP Address – Reboot required

If a Valcom IP Gateway, Speaker, Server or other endpoint with a static IP address periodically requires a reboot, but always has LEDs illuminated (indicating that it is still powered), then temporarily route the network connection through a small unmanaged PoE or PoE+ switch[1]. The reason for this step is to facilitate testing independently of the network when the trouble occurs, without rebooting the PoE endpoint.

If the trouble returns, then disconnect the network from the unmanaged PoE or PoE+ switch and test the endpoint per the "How to Test Basic Functionality" section of this manual. If the endpoint fails the basic functionality test, then cycle power and test again. If, at this point, cycling power restores endpoint operation, then contact technical support.

[1]Unplug the endpoint from the network, connect it to the unmanaged PoE or PoE+ switch, connect a second patch cable from the unmanaged PoE or PoE+ switch to the endpoint's original network port.

Disconnect Troubles

Disconnect troubles may be:

a) The Valcom system does not release after the page is complete

1. If access is via a telephone system FXO Port, Trunk Port or C.O. Line Port, then temporarily replace the phone system with your lineman's handset and try to reproduce the trouble. If you cannot, then verify the operation, type and programming of the phone system's port. The FXO Port, Trunk Port or C.O. Line Port should operate just like your lineman's handset.

2. If access is via a telephone system station port into a Valcom FXO Gateway, then verify that the station port is providing proper disconnect signaling (Open Loop Disconnect), or, if using silence detection, use your lineman's handset in monitor mode (Hi-z) to verify that the station port is completely silent after the calling party disconnects.

b) The Valcom system disconnects during an announcement

1. If access is via a telephone system FXO Port, Trunk Port or C.O. Line Port, then temporarily replace the phone system with your lineman's handset and try to reproduce the trouble. If you cannot, then verify the operation, type and programming of the phone system's port. The FXO Port, Trunk Port or C.O. Line Port should operate just like your lineman's handset. Note that intermittent wire connections may be the culprit.

2. If access is via a telephone system station port into a Valcom FXO Gateway, then verify (directly on the FXO Gateway Input) that the station port is providing constant voltage for the duration of the connection, that pauses in audio are not exceeding any silence disconnect settings and that there is not an absolute timeout value set in the FXO Gateway.

3. If access is via SIP, verify that the SIP server does not have any absolute timeout setting enabled. You can do this by monitoring the Paging Diagnostics screen in the VIP-102B IP Solutions Setup Tool during a group or all call announcement. If a Page Stop command is consistently issued for group announcements mid-stream, then the SIP paging Server settings should be investigated.

c) In poor designs, power supplies or PoE ports may enter protection mode during announcements. If overloading is the trouble, then disconnect symptoms will typically be isolated to specific PoE switches, PoE ports or to specific power supplies. Additionally, with analog systems, if the main head end equipment is

inadequately powered, then the disconnect will typically manifest itself during long all call announcements. Verify system loads on PoE switches and power supplies with your [PoE load](#) and [Volt Ohm meters](#). Measure both before and during the disconnect sequence under investigation.

d) We've experienced issues where network endpoint monitoring programs that ping a network's endpoints can cause IP speakers to disconnect in the middle of an announcement. This occurs when the ping rate is set abnormally high. This activity utilizes a significant amount of the speaker's microprocessor power and causes interruptions in audio.

Intermittent Troubles

Intermittent troubles, troubles that seem to occur at random times and are not reliably reproduced, can be difficult to troubleshoot. Mostly because *if you cannot reliably reproduce the trouble, then how will you know if you've resolved it*? These troubles may be user related, environment related, design related, network related, equipment related or any combination of these. Establishing a pattern of when and where the troubles occur, the users involved, and what changes in the environment while the trouble manifests itself often leads to resolution.

With any equipment, if your install has multiples of any device, and only some exhibit troubles, then swapping the ones that work reliably with the ones exhibiting intermittent troubles can help lead you to a conclusion. [Log files](#) may be used to document system activity when intermittent troubles occur.

Relocate the equipment for testing to help eliminate environmental variables. Environmental variables, to name a few, includes the ambient temperature of the mounting location, noisy "dirty" ac voltage, EMF from surrounding equipment and radio frequency interference. If the trouble ceases when the equipment is initially powered, but fails after time, it may be an internal thermal issue.

For networked equipment, poorly terminated RJ45s, 110 blocks and/or variances in network traffic can cause issues. Heavy network traffic can result in broken or missed audio and unreliable system operation. While VLANs do not increase the available bandwidth of a network, they can restrict traffic only to ports where it is intended. Therefore, Valcom always highly recommends a dedicated VLAN, or a VLAN shared with

the telephone system. If rebooting the endpoint temporarily solves the trouble, then improper network setup is possibly the cause.

Improper or incomplete multicast setup is always a suspect when individual audio channels work, but group audio fails. Dependent upon network design, Valcom VoIP devices may randomly fall in and out of the multicast group required for group audio.

For example, if successful one-way audio distribution, (group announcements, tones, music initiation, etc.) requires **multiple initiation attempts** after being idle for some period of time, however, once successfully initiated, works consistently until once again idle for some period of time, the issue may be that once the network multicast source-to-receiver pathways have timed out and have been deleted, it is taking several multicast packets to rebuild the path.

One possible cause could be the number of packets required by the network router(s) before the multipath routes are reestablished. For whatever reason, the network may absorb several packets before the following packets would be delivered to the endpoints. If this cannot be corrected in the network configuration, then firmware to add additional "page start" message packets may be required for the Valcom IP devices.

The IP6000 Initial Setup Procedure clearly outlines the network requirements for Valcom IP6000 IP based solutions. For standard Valcom VoIP installations, network requirements may be found here.

Valcom's VIP status monitor may be utilized to monitor Valcom VoIP devices and report if they no longer respond to a ping and may help to identify intermittent connectivity.

Other factors that can contribute to intermittent troubles include:

a) Mounting locations that exceed the equipment's rated operating temperature and humidity
b) Intermittent wiring connections/poorly crimped/loose connectors/short circuits
c) Wrong programming (example - making scheduled tones lower priority than announcements will result in missed scheduled tones)
d) Excessive load on power supplies, PoE switch ports or audio outputs
e) AC Switches of circuit breakers being toggled
f) Improper equipment grounding

g) Failure to release the system after an announcement (pressing hold instead of release)

h) Failure to program using the most current version of programming software

i) IP address conflicts with PCs or other endpoints that are powered on (conflict occurs) and off (conflict goes away) as needed

h) Periodic variances in ac voltage

i) People

People? Several years ago, a customer who had a Valcom IP system installed also had endpoint monitoring set up. Occasionally, on random evenings, endpoints in different parts of the facility would disappear from the network for 20 or 30 minutes and then reappear. Attempts to find the cause proved elusive.

Finally, the network admin was working late one evening when the trouble occurred. She visited the area where the endpoints were unresponsive only to find that the evening cleaning crew, needing an outlet for the vacuum, unplugged the local network switch. Once vacuuming was complete, the switch was reenergized, and the endpoints came back online.

Clock Correction Troubleshooting

If Valcom clocks are displaying incorrect time, then step 1 is to make certain that the [time source](#) (master clock, controlling intercom, NTP server) is set to the correct time.

Several Valcom intercom products can correct secondary clocks. In addition, Valcom sells a wireless clock system that can be integrated with many of our intercoms or may be used stand alone.

There are basically 4 categories of clock correction:

      a) Wireless
      b) Corrected by relay closures (synchronous, hourly, impulse, etc.)
      c) Digitally corrected
      d) Corrected over Ethernet via NTP

Wireless clocks, as the name implies, obtain time correction information via a wireless signal. This signal may be received directly from the transmitter or repeater or may come from adjacent clocks as they transceive. In order to test an analog display wireless clock, there are numerous built in diagnostic tests that may be performed to test battery strength, signal strength, time of last correction and mechanical integrity. The analog display wireless clock manuals will describe how to invoke these diagnostic tests.

If you apply power to a wireless clock while it's close to the main transmitter or repeater, it should obtain correction signal and display the same time as the transmitter or the system sourcing the repeater. If it does, then the clock is functioning properly. If the clock functions properly per the previous test but will not obtain correct time in its intended location (the process could take a couple of days), then perhaps it is out of range of a transmitting or transceiving device.

Performing a wireless [clock site survey](#) will determine clock signal coverage in a facility. If an analog wireless clock attempts correction, however, displays an incorrect time, then clock diagnostics, as described in the manual, should be utilized to detect any hardware issues. Wireless clocks typically perform well when <=60' spacing is used.

Another possible cause of wireless clocks displaying different or incorrect times might be that they are receiving time from multiple wireless master clocks (the intended master and a master clock in a neighboring facility). To test for this, unplug the known master clock and cycle power to a wireless clock. It if obtains time then there is still a wireless master transmitting somewhere close by.

Wired analog display clocks also have built in diagnostic tests for protocol verification, comprehensive clock testing and to set manufacturer defaults. The procedures for these tests are described in the installation manuals.

As far as **clocks corrected by relay closure**, the correct wiring and sequence of closures is required for successful correction. Once you've determined the proper correction protocol, generic wiring diagrams for each can usually be found in the Valcom intercom manual.

The Valcom intercom simply closes relays on the V-CIO Clock Relay Card. These relays close at times based upon the correction protocol selected in the intercom.

Determining if the Valcom system is properly closing the clock correction relays is as simple as using your Volt Ohm meter to measure for continuity between each relay's common and normally open contact at the times dictated by the correction protocol.

This test may be expedited by, temporarily disconnecting all wires from the V-CIO output relays, changing the system time to 5 minutes before the correction is supposed to occur, and verifying that the relays close in the proper sequence. If the relays are closing at the correct times and for the correct duration*, then the Valcom equipment is working properly.*

Connecting a single clock in the equipment room (all other clock cable runs disconnected) is a good way to determine if the head end equipment is properly configured and the correct correction protocol has been selected, without introducing clock cabling variables. Note that clock correction by relay closure may take several days to correct the clocks.

If ac clocks periodically jump an hour ahead in time, or if ac clocks will not correct properly, then resistors R5, R6, R7 and R8 of the V-CIO must be removed.

Valcom Digitally corrected clocks (2-wire digital correction) are corrected over a single pair of UTP wiring. The devices that typically source the clocks are a V-DCPI Digital Clock Protocol Interface and a V-CCU* Clock Controller.

The V-DCPI converts correction data from the Valcom Intercom into RS485.

The V-CCU* mixes power and data on a single pair of wires and sends it to the clocks.

In proper operation, the green data LED of the V-DCPI will frequently flicker as it receives and processes the *polarity sensitive* correction data from the intercom. If that correction data is originating from a V-SER clock card in an analog intercom, then the V-SER will display rapidly flashing LEDs as well.

The V-CCU*'s red data error LEDS will not be illuminated and the green output data LEDs will flash every time correction data is sent to the clocks. The rate of correction is determined by dipswitches and is typically set to once a second.

If the V-CCU*'s red data Error LEDS are on, then the V-CCU* is not receiving input data or the power supply is not properly connected.

If the V-DCPI's green Data LED is not flashing following the selected correction interval*, then it is not receiving data from the intercom, it is not properly connected to the V-CCU** or the intercom is not set to send data in a Digital Out/24-Hour Enhanced Mode. The V-DCPI does not require a separate power supply when used with the V-CCU.

In some systems, a master clock will be used in place of the V-DCPI.

The V-CCU* must be power cycled before dipswitch changes are recognized.

There are limits for how many clocks can be sourced from each of the V-CCU's outputs. Refer to the V-CCU* Manual for details. There are also limits for clock quantities per cable run bases upon type, wire gauge and distance. Refer to the following chart:

Using 24 AWG UTP for 24V Clocks					
	1 Clock per Wire Run	5 Clocks per Wire Run	10 Clocks per Wire Run	15 Clocks per Wire Run	20 Clocks per Wire Run
Analog	3000'/914 m	1000'/304 m	600'/183 m	400'/121 m	300'/91 m
2.5-inch Digital	1500'/457 m	500'/152 m	250'/76 m	N/A	N/A
4-inch Digital	400'/121 m	100'/30 m	N/A	N/A	N/A

Using 20 AWG UTP for 24V Clocks					
	1 Clock per Wire Run	5 Clocks per Wire Run	10 Clocks per Wire Run	15 Clocks per Wire Run	20 Clocks per Wire Run
Analog	7000'/2130 m	2500'/762 m	1400'/427 m	1000'/304 m	750'/229 m
2.5-inch Digital	3900'/1189 m	1300'/396 m	700'/213 m	N/A	N/A
4-inch Digital	900'/274 m	300'/91 m	N/A	N/A	N/A

*Correction interval may be set on the V-CCU's dipswitches or may be set on the main data source – master clock, VECPU6, etc. If set on the main data source, then the V-CCU dipswitches should be set to send at the once a second rate or set to slave mode. Slave mode simply sets the V-CCU to once a second and disables the red data error LEDs.

**The V-CCU is part of the V-VCU package. The V-VCU contains a V-VCU and a power supply.

Both the V-CCU and the power supply have independent over current protection.

If the green LED output for any properly loaded (<=40 VPU) V-CIO clock circuits turns off when clocks are connected, then remove the clock circuit wiring from the V-CCU.

Check to be certain that the power supply output for that clock circuit is providing 24vdc within 60 second of removing the clock circuit wiring. If not, retest the power supply output with all wiring removed to eliminate wiring issues. If the power supply does not provide voltage output with no wiring connected, and the power supply is connected to a known live ac outlet, then replace it.

If the power supply tests good, then the V-CIO circuit under test's green LED should re-illuminate within 60 second of removing the clock circuit wiring. If not, the unloaded V-CIO circuit is suspect.

If the V-CIO circuit under test's green LED does re-illuminate within 60 second of removing the clock circuit wiring, then measure the resistance across the disconnected circuit's wire pair, and from each of the 2 conductors to power supply ground, to determine if there are any short circuits. If these continuity tests reveal short circuits, then they must be corrected before proceeding.

Once it has been determined that there are no short circuits, then disconnect all clocks, reconnect the circuit wiring to the V-CCU, and verify that the green data associated with the circuit is illuminated. Reconnect the clocks one at a time to determine the point at which the circuit becomes overloaded – i.e. green LED diminishes. Once you've determined that a clock is loading down the clock circuit, disconnect it, allow the green LED to re-illuminate and continue reattaching clocks one at a time.

IP Clocks connect to a properly configured network and obtain time from an NTP server. If they do not correct, ensure that the network is properly configured, the PoE switch power budget has not been exceeded and that the port used for the clock is routable to a valid NTP source.

Note that some network DHCP servers will restrict access to the Internet for endpoints that are set for static addresses. This can be confusing when the NTP server that you are using is Internet based. Tests for Internet access from a switchport are usually accomplished with a PC. If the PC is using DHCP it may be able to access an Internet based NTP server while a statically addressed IP clock cannot.

Testing Routing

Use the "ping" command to test basic TCP/IP routing from one network location to another.

The ping command is a Command Prompt command used to test the ability of a source computer to reach a specified destination computer. The ping command is usually used as a simple way verify that a computer can communicate over the network with another computer or network device. Refer to http://www.wikihow.com/Ping-an-IP-Address.

If your source is not a computer, you can test routing by configuring your laptop NIC to use the same IP address as the endpoint in question and replacing the endpoint (use the same switch port) with your laptop. Try to ping the IP address of the destination under evaluation. If unsuccessful then the switch port is not routable to the destination (this could be as simple as an invalid gateway address in the source device).

Replacing an endpoint with your laptop will not work if switch ports are locked down by MAC addresses.

If pinging a domain name fails but pinging the resolved IP address of the domain name works, then your DNS server entries should be evaluated.

The following is credited to https://technet.microsoft.com:

Destination Host Unreachable

This message indicates one of two problems: either the local system has no route to the desired destination, or a remote router reports that it has no route to the destination. The two problems can be distinguished by the form of the message. If the message is simply "Destination Host Unreachable," then there is no route from the local system, and the packets to be sent were never put on the wire. Use the Route utility to check the local routing table.

If the message is "Reply From < IP address >: Destination Host Unreachable," then the routing problem occurred at a remote router, whose address is indicated by the "< IP address >" field. Use the appropriate utility or facility to check the IP routing table of the router assigned the IP address of < IP address >.

If you pinged using an IP address, retry it with a host name to ensure that the IP address you tried is correct.

Request Timed Out

This message indicates that no Echo Reply messages were received within the default time of 1 second. This can be due to many different causes; the most common include

network congestion, failure of the ARP request, packet filtering, routing error, or a silent discard. Most often, it means that a route back to the sending host has failed. This might be because the destination host does not know the route back to the sending host, or one of the intermediary routers does not know the route back, or even that the destination host's default gateway does not know the route back. Check the routing table of the destination host to see whether it has a route to the sending host before checking tables at the routers.

For most basic tests, the ping command will not require additional syntax. The following detailed information is included for reference.

The following is credited to http://pcsupport.about.com/:

The ping command operates by sending *Internet Control Message Protocol (ICMP) Echo Request* messages to the destination computer and waiting for a response. How many of those responses are returned, and how long it takes for them to return, are the two major pieces of information that the ping command provides.

Ping Command Syntax

ping [**-t**] [**-a**] [**-n** *count*] [**-l** *size*] [**-f**] [**-i** *TTL*] [**-v** *TOS*] [**-r** *count*] [**-s** *count*] [**-w** *timeout*] [**-R**] [**-S** *srcaddr*] [**-p**] [**-4**] [**-6**] *target* [**/?**]

Tip: See How To Read Command Syntax if you're not sure how to interpret the ping command syntax above.

-t = Using this option will ping the *target* until you force it to stop using Ctrl-C.
-a = This ping command option will resolve, if possible, the hostname of an IP address *target*.
-n *count* = This option sets the number of ICMP Echo Request messages to send. If you execute the ping command without this option, four requests will be sent.
-l *size* = Use this option to set the size, in bytes, of the echo request packet from 32 to 65,527. The ping command will send a 32-byte echo request if you don't use the **-l** option.
-f = Use this ping command option to prevent ICMP Echo Requests from being fragmented by routers between you and the *target*. The **-f** option is most often used to troubleshoot Path Maximum Transmission Unit (PMTU) issues.
-i *TTL* = This option sets the Time to Live (TTL) value, the maximum of which is 255.
-v *TOS* = This option allows you to set a Type of Service (TOS) value. Beginning in Windows 7, this option no longer functions but still exists for compatibility reasons.

-r *count* = Use this ping command option to specify the number of [hops](#) between your computer and the *target* computer or device that you'd like to be recorded and displayed. The maximum value for *count* is 9 so use the [tracert command](#) instead if you're interested in viewing all hops between two devices.

-s *count* = Use this option to report the time, in Internet Timestamp format, that each echo request is received and echo reply is sent. The maximum value for *count* is 4 meaning that only the first four hops can be time stamped.

-w *timeout* = Specifying a *timeout* value when executing the ping command adjusts the amount of time, in milliseconds, that ping waits for each reply. If you don't use the **-w** option, the default timeout value is used which is 4000, or 4 seconds.

-R = This option tells the ping command to trace the round-trip path.

-S *srcaddr* = Use this option to specify the source address.

-p = Use this switch to ping a *Hyper-V Network Virtualization* provider address.

-4 = This forces the ping command to use IPv4 only but is only necessary if *target* is a hostname and not an IP address.

-6 = This forces the ping command to use IPv6 only but as with the **-4** option, is only necessary when pinging a hostname.

target = This is the destination you wish to ping, either an IP address or a hostname.

/? = Use the help switch with the ping command to show detailed help about the command's several options.

Note: The **-f**, **-v**, **-r**, **-s**, **-j**, and **-k** options work when pinging IPv4 addresses only. The **-R** and **-S** options only work with IPv6.

Other less commonly used switches for the ping command exist including [**-j** *host-list*], [**-k** *host-list*], and [**-c** *compartment*]. Execute **ping /?** from the Command Prompt for more information on these two options.

References:

http://www.wikihow.com/Ping-an-IP-Address

https://technet.microsoft.com/en-us/library/cc940095.aspx

[Pinging from within a Valcom IP Gateway with the VIP Utility Tool](#)

[VIP Utility Tool](#)

Testing Multicast

When working with Valcom IP based systems, a proper installation includes maintenance of multicast forwarding tables using IGMP querying (preferred), IGMP snooping, a combination of both, or similar functions. Improper configuration of multicast is one of the most common issues encountered when deploying a Valcom IP Solution and can result in the inability to discover devices on the network and in the inability to perform such routine tasks as group announcements. The IP6000 Initial Setup Procedure and Valcom VoIP Initial Setup Procedure clearly outlines the network requirements for Valcom IP6000 IP based solutions.

Should troubleshooting be required, a multicast diagnostic tool is available at www.valcom.com/esd. This software simultaneously sends and receives multicast packets to/from up to 5 multicast addresses. Installing this program on 2 PCs allows verifying multicast traffic between 2 network switch ports.

The tool is very easy to use:

j) Install the VIP-102B IP Solutions Setup Tool _and_ the Valcom IP Solutions Multicast Diagnostic Tool on each PC that will be used for testing. Disable all firewalls and wireless network interface cards on these PCs for the duration of the testing. Note that on some PCs turning off the Wi-Fi from the system tray doesn't always turn it off on the backend, it may have to be disabled in the Services menu.
k) Exit all other Valcom programs (like the VIP-102B)
l) Assign the PCs valid IP addresses for your VLAN and plug the PCs into the VLAN network ports under evaluation.
m) Set the send interval to the same value on all computers involved in the test.
n) Check "Echo Received Packets by Multicast", "Echo Received Packets by Unicast", "Receive All" and "Start" on one or more "remote computers"
o) Check "Send all" and "Receive all" and "Start" on a "control computer"
p) Allow the control computer to receive 50 or more packets from each remote computer.
q) Click "Stop", then click "Details" on the control computer to verify reception of data and see calculated average round trip times from each remote computer.

This software has been run on Windows XP, 7, 8.1 and 10, but has not been tested extensively.

Unique address/port are required for accuracy (For example, setting multiple multicast addresses to use the same port will probably not return accurate counts. The total

number of packets received may be correct, but it is indeterminate which particular thread will get the delivered packet)

There is a performance penalty during startup that will cause the packet delivery time to be inaccurate. Generally, the first packet or two will show a longer RTT (Round Trip Time) due to startup delays. The average RTT becomes more realistic after running for some bit of time. Allow time for 50 or more packets to be received for accurate results.

The round-trip time also includes the time used to process the packets by the program itself, which adds some small amount to the total. If RTT is more than 40 milliseconds, which would be 20 milliseconds one-way, then VoIP audio could be affected.

Typically, one "control computer" will be set to "Send All" and "Receive All" but not to "Echo Packets Received by Multicast" or to "Include Unicast Response to Sender".

Other "remote computers" connected to different network ports under test will be set to "Receive All", "Echo Packets Received by Multicast" and to "Include Unicast Response to Sender", but not to send anywhere.
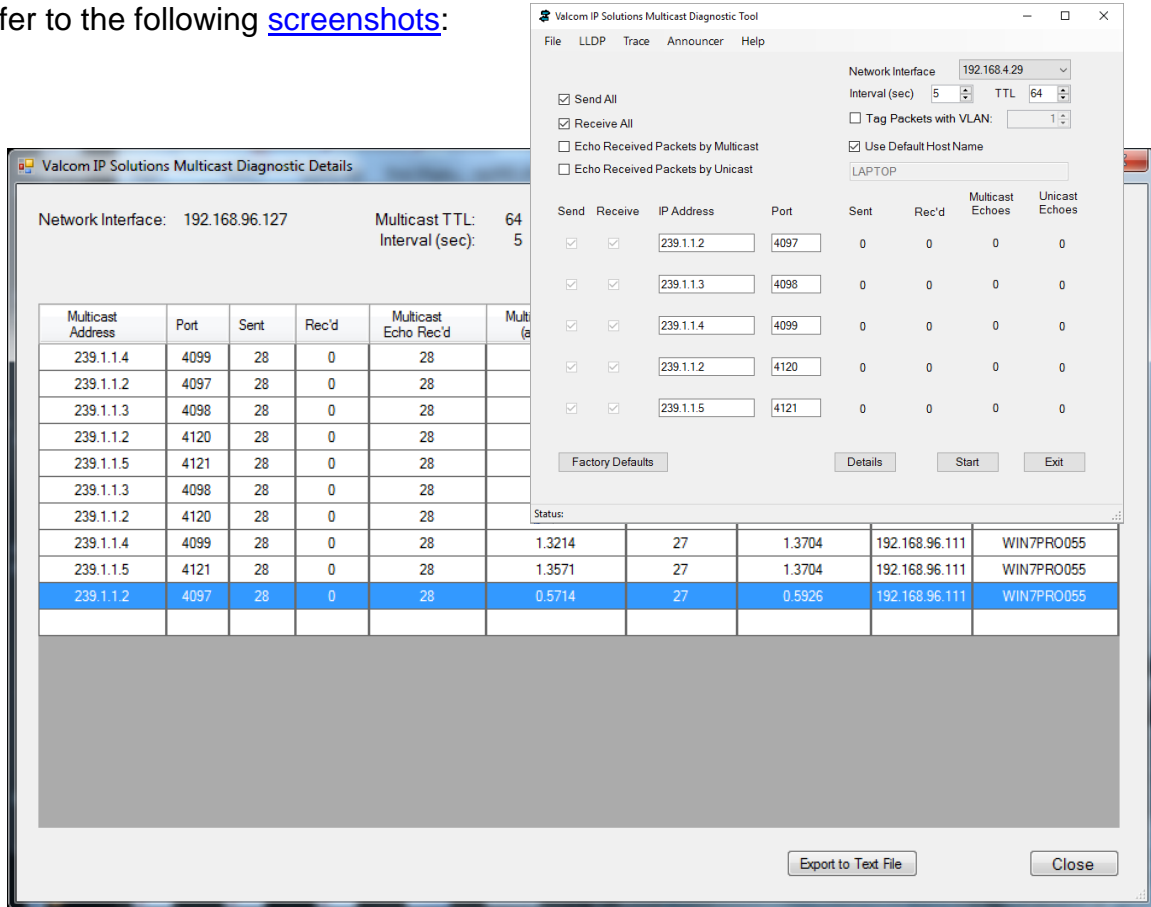
This will allow the "control computer" to test multicast traffic to and from each "remote computer" and to calculate average RTT from each.

If unicast echoes back from a remote computer, but multicast does not, then the remote computer received multicast from the control computer but was unable to send multicast back.

If neither multicast nor unicast echo back from a remote computer, then the remote computer did not receive multicast from the control computer or was unable to send multicast or unicast back.

Note that for scenarios where cycling the network connection temporarily cures network issues, waiting until the trouble occurs to plug in the testing computers could skew the test as they may initially join the multicast group when they are connected to the Ethernet. In this case, prepping the test by connecting the test computers and then waiting until the network failure occurs to invoke MultiCastDiags may be necessary.

Refer to the following screenshots:

Tip – Prove that the PCs running MultiCastDiags.exe don't have any blocked ports that will skew the test results. Initially test with the two PCs connected to a simple unmanaged switch or through a cross over cable. If the multicast testing is successful   on this "bench test" but not on the actual network, then you can be certain that the network is not properly configured.

If successful paging is sporadic,

For example:

The first page is unsuccessful
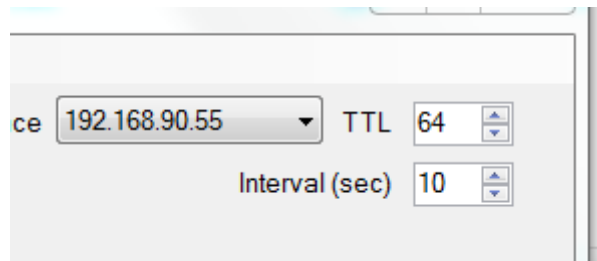
The second page is also unsuccessful

The third page is successful and subsequent page attempts immediately following the successful page continue to work

However, if you wait for some period of time, the **failure sequence** starts over again.

(or some similar pattern)

Then it is likely that the problem is that the multicast membership in the remote subnets is not being communicated up through the network to the originating subnet. Each page opens a multicast stream (first the control, then the audio) and it all times out after some period of inactivity.

To test with Multicastdiags.exe, Start with the interval set to a value just above the time required to reinitiate the **failure sequence**, then change the interval to 5 or 10 seconds and compare the results.

If the multicast test is consistently successful at the shorter interval, but fails at the longer interval then the multicast membership in the remote subnets is not being communicated up through the network to the originating subnet.

Other MultiCastDiags Features:

Multicast Trace will query routers to find out how multicast would be handled from one point to another. This is dependent on the routers having the mtrace function built-in.

LLDP will send an LLDP packet from the PC into the switch. If LLDP is turned on in the switch, then some information from the switch may be returned. Interesting information could be things such as the port's VLAN assignment, whether Voice VLANs are in use (and which VLANS are Voice VLANs), the switch port designation, etc. LLDP uses the WinPCap library, which is installed with Wireshark. WinPCap can be installed separately, but it is easier to install Wireshark and let it do the install.

MAC/IP Announcer will send the MAC and/or IP address announcement as a unicast page to an individual device. This may be useful for locating speakers that were installed without properly documenting their location.

A simpler, but less comprehensive test of multicast can be performed with the VIP-102B IP Solutions Setup Tool.

If the PC is properly configured as an endpoint on the Valcom VLAN then Communications/Network Diagnostics should display green indicators for both the broadcast and multicast beacons that are generated by each Valcom IP endpoint. The multicast beacon is sent using the "control" multicast address. (note: opening the VIP-102B sends a multicast "join" message every time the Communications/Network Diagnostics screen is invoked).

You can also use the Communications/Paging Diagnostics page utility of the VIP-102B to verify that page start and page stop commands are being sent when groups are dialed and when the group page is concluded.

If multicast is properly configured, then the Valcom IP speakers, LED signs, gateways and servers will join the multicast group when they are connected. If you have a situation where IP endpoints periodically stops broadcasting one-way group audio, or other multicast-initiated functions cease on an endpoint until power is cycled, the endpoints may be falling out of the multicast group. If your network relies upon IGMP snooping to maintain multicast membership, and your system is idle long enough, meaning no multicast traffic for a given length of time, then the IGMP snooping may remove the device switch ports from the multicast group. If your system has an Application Server, creating a daily schedule to send audio to an empty group every 10 minutes or so will create the traffic that the IGMP snooping needs to maintain membership. Or use IGMP querying along with, or instead of, IGMP snooping. The fact that the speakers are still able to operate as 2-way intercoms (individual dial codes) during the one-way audio failure, is a very strong indicator that they are not consistently receiving multicast network traffic. The speakers utilize UTP for 2-way intercom calls and multicast for all one-way audio (all call, group announcements, bells).

Device Multicast settings

It's important to note that the control, setup, debug and audio multicast addresses for all the Valcom servers, gateways and endpoints can be changed, however, the values for equipment intended to work as a system must be the same.

<u>Troubleshooting multicast related random loss of one-way audio</u>
If rebooting (power cycling) the VoIP endpoints temporarily clears the trouble, it is most likely because the endpoints will request to join the multicast group upon start up. After that, the network manages which endpoints will continue to receive multicast traffic and which will not. If the network is properly configured, then the VoIP endpoints will always receive multicast as intended.

To prove that endpoints are temporarily losing access to multicast traffic, a 4-minute Wireshark data capture will be required during the failure mode. The 4-minute length is to ensure that any IGMP join messages are captured. Note that this process could temporarily fix the problem (for example if the endpoint being tested is rebooted as part of the test set up) so great care must be taken to preserve the failure state until data can be captured.

While in the failure state, the network administrator will need to mirror a port of one of the endpoints exhibiting the trouble. They will then need to perform a 4-minute Wireshark

---

capture of the data received at the mirrored port during both a group page or all call that would normally include the speaker under test, and during an individual channel call to the endpoint under test.

For comparison, a second 4-minute Wireshark capture should be performed when the endpoint is working properly (receiving one-way audio audio).

Following this, the network administrator will need to analyze the packet capture to ascertain if the port (and therefore the endpoint under test) is receiving the multicast traffic (Valcom can assist with this analysis once the data is captured).

Troubleshooting [Service Location Protocol](#) and DHCP

The Valcom **[SLP-Responder](#)** Utility program is intended to help diagnose and resolve problems related to Service Location Protocol (SLP) and Valcom IP Speakers. By providing an application to easily allow responses to SLP requests sent by Valcom speakers when booting up, this program may help with diagnosing SLP multicast network problems.

Service Location Protocol is designed to use multicast address 239.255.255.253 and UDP ports 427 (listener) and 437 (sender). This utility will listen for SLP requests from Valcom IP Speakers and respond. Only requests for "InformaCastConfiguration" in the "Berbee® Applications" scope are recognized and responded to.

When using this program to serve as a substitute SLP responder, the configuration will generally use this computer's network interface for the SLP IP Address field and the IP address of the InformaCast® server as the Register IP Address. The default port (8081) will usually be correct but may be changed if necessary.

When used this way, the local PC should be in the same subnet with the Valcom Speaker being diagnosed. This will provide the SLP service close to the Speaker. If the Speaker can register with the InformaCast® server after getting SLP information from this PC, then this would indicate a network problem with multicast.

This program may also be used to allow Valcom InformaCast® Speakers to be booted up in an environment where there are no InformaCast® servers. Typically, this can be used for situations where an InformaCast® speaker was ordered incorrectly or for troubleshooting IC speakers that have been removed from the IC environment.
For this use, both the SLP and Register IP address fields would be set to use this PC's IP address. This program will then respond to the SLP requests and provide the

configuration file to the speaker. This information allows the speaker to boot up normally, after which it can be scanned and configured using the VIP-102B IP Solutions Setup tool.

A minimal DHCP server is provided for use on networks where DHCP service is not already available, or for troubleshooting connectivity from the normal DHCP service. To enable this feature, check the box to Provide DHCP Service and enter the DHCP scope parameters.

It is important to keep this program running until all speakers have been configured as necessary.

This program requires Administrator privileges to run properly. When starting the program, use the right-click menu option to "Run as Administrator". On the first run, Windows Firewall may prompt for permission to allow communication through the firewall. This must be agreed to for this program to work.

Additionally, it may be required to manually add a firewall rule to allow HTTP traffic on the port selected as the Register Port (8081 by default). This is typically done through the Control Panel -> Windows Firewall applet. Under "Inbound Rules", use the New Rule wizard to allow TCP connections on the port being used.

DHCP access restriction

Note that some network DHCP servers will refuse to issue network credentials at all unless the endpoint's MAC address is white listed.

In addition, some network DHCP servers will restrict access to the Internet for endpoints that are set for static addresses. This can be confusing as tests for Internet access are usually accomplished with a PC, which may be using DHCP, while the actual IP endpoint under evaluation is not.

Assessing Field Wiring

Audio field wiring should be accessed before connections are made to the source equipment. This is especially true in retrofit situations.
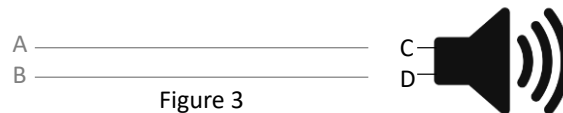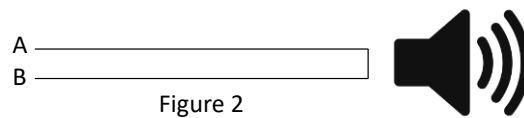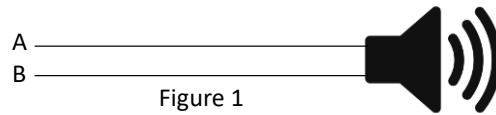
Referring to Figure 1, with points A and B disconnected from equipment,

a) Use your Volt Ohm meter to verify that there is no ac or dc voltage between point A and B. If there is, replace or troubleshoot the cable.

b) Use your Volt Ohm meter to verify that there is no ac or dc voltage between point A and equipment ground or Earth Ground. If there is, replace or troubleshoot the cable.

c) Use your Volt Ohm meter to verify that there is no <u>continuity</u> between point A and equipment ground or Earth Ground. If there is, replace or troubleshoot the cable.

d) Use your Volt Ohm meter to verify that there is no <u>continuity</u> between point B and equipment ground or Earth Ground. If there is, replace or troubleshoot the cable.

e) Use your [impedance meter](#) to measure across points A and B to verify that the impedance of the speaker line is suitable for the audio source being used. Refer to the chart below:

    1) If the results indicate a higher than expected impedance, or open circuit, then disconnect the speaker and short the cable pair as shown in Figure 2. Repeat Step e)

        i. If lower impedance is measured then, referring to Figure 3, measure across point C and D to determine if the speaker coil has abnormally high impedance. If so, then replace the speaker

        ii. If there is little or no change, then replace or troubleshoot the cable.

2) If the results indicate a lower than expected impedance or short circuit then, referring to Figure 3, measure across point C and D to determine if the speaker coil has abnormally low impedance. If so, then replace the speaker, otherwise replace or troubleshoot the cable.

Wire pairs used for voltage should also free of short circuits and continuity to audio wiring.

Expected Speaker Line Impedance

Speaker Type	Number of speakers	Sum of Tap settings	Expected Impedance Reading
45 Ohm	1	N/A	=> 45 Ohms
45 Ohm	2	N/A	=> 22.5 Ohms
25 Volt	Any	<= 1 Watt	=> 625 Ohms
25 Volt	Any	<= 5 Watt	=> 125 Ohms

System Audio Level Balancing

Analog Systems

Public address speaker systems generally have one or more means of volume adjustment. Old fashioned centrally amplified (25v,70.7v, 100v) speakers generally have multiple connection wires called taps. The audio cable from the amplifier connects to one tap (common + a color-coded wire indicative of the wattage level that will theoretically result from applying the rated amplifier signal). Adjusting these taps after installation *may* require that the speaker be removed, adjusted, and then reinstalled.
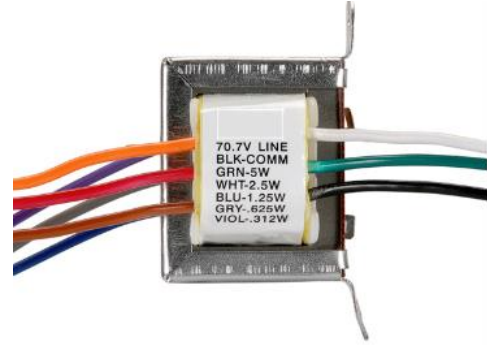
In some cases, the amplifier or other audio source provides a fixed output level and in some cases the output level is adjustable. If the amplifier provides a fixed output level, consideration should be given to adding a suitable inline volume adjustment.

Self-amplified speakers feature easy-to-use integrated volume controls and are often sourced by a "page control" or other audio source. In some cases, the "page control" or other audio source provides a fixed output level and in some cases the output level is adjustable.

Regardless of which speakers are utilized, it is generally advisable to adjust or tap all speakers in an area to the same, typically mid-range, volume. Also adjust any inline volume controls to a mid-range setting.

Once accomplished, adjusting all the speakers in an area may be accomplished by adjusting the source, or if the source output level is not adjustable, via a compatible inline volume control.

Note: It's important, that speakers be wired in phase, meaning that the same connection point on all speakers be connected to the same connection point on the source, and that all speakers and horns in an area be directed in a common direction.

Initial Audio Level Setup

Connect a music source or other repeating audio source such that the audio is broadcast across all speakers and horns at a level representative of a typical voice page[1], walk through the facility and verify that each speaker is receiving audio.

Make area specific volume adjustments as necessary.

If talkback speakers and call buttons are installed, verify the operation of each talkback speaker by pressing its associated call button and conversing. Make area specific volume adjustments as required. (some talkback speakers have volume controls, some have taps, some do not feature "per speaker" volume adjustment however, may have an inline volume control).

If call buttons are not installed, verify the operation of each VoIP talkback speaker by calling each speaker and conversing. Make speaker specific volume adjustments as required.

Call each zone (individual speaker circuit) via the system telephone (or other intended access method) and verify that audio level is adequate, and that audio is clear. (Note that distorted audio will result from excessive audio level being applied to an amplifier, self-amplified speaker or page control input.)

Involve the owner in the final testing to facilitate acceptance.

By setting all the speakers in an area to the same volume (tap setting or volume control) you achieve a baseline volume for each area.

This baseline volume will facilitate the ability to maintain balance regardless of the audio source. If any applied source is too loud or too quiet, adjustment can be made via the source without the need to make speaker level adjustments.

VoIP Systems

Valcom VoIP systems have many volume adjustments. In this section we will initially discuss system volume balancing. We will be discussing individual channel output volume adjustments, offsets that may be applied to group announcements as well as offsets that may be applied to Application Server events.

First, there are preset output volume adjustments per endpoint channel. These presets dictate the broadcast level when the channels are accessed via their individual channel dial codes.
Note: It's important that all speakers and horns in an area be directed in a common direction.

These preset output volume adjustments may be increased or decreased (offset) when the channels receive audio as part of a group.

VIP-102B Group Priorities vs. Server Group Priorities

The VIP-102B IP Solutions Setup Tool allows volume offsets to group audio based upon the group's priority. If the group is receiving audio from an Application Server event, then the group's priority and its associated volume offset is overridden by the event's priority and volume offset.

For example, an announcement to a group that has an assigned volume offset of +6 will result in the group members broadcasting the audio with an offset of 6 above their individual channel output volume presets.

An Application Server sending an audio event with an assigned volume offset of +2 to a group with an assigned volume offset of +6 will result in the group member broadcasting their audio with an offset of 2 above their individual channel output volume presets.

In addition, gateways may be providing audio to analog subsystems featuring their own volume adjustments.

Initial Audio Level Setup

The following important process is intended to determine baseline volume levels for your VoIP speakers and audio gateways. The process will facilitate system balance by determining and setting baseline levels first and then adjusting individual area, group or event volumes up or down as necessary.

First, make a test announcement to All Call or any one-way group comprised of VoIP One Way speakers. Be certain that the group being used for testing has a priority volume offset of zero. Determine an acceptable volume level.

Using the "Programming/Volume Adjustments" section of the VIP-102B, adjust the channel output volume of all VoIP One Way speakers to the level determined above.

Secondly, make a test announcement to a VoIP Talkback speaker that is installed in a typical area (an area representing the location of most of your talkback speakers). Determine an acceptable volume level.

Using the "Programming/Volume Adjustments" section of the VIP-102B, adjust the channel output volume of all VoIP Talkback speakers to the level determined above.

For this next adjustment, you will need for an assistant to be in an area representative of your typical VoIP Talkback speaker locations. Call into the selected area and converse in a normal level/tone of voice. Determine an acceptable receive (input) volume level.

Using the "Programming/Volume Adjustments" section of the VIP-102B, adjust the channel input volume of all VoIP Talkback speakers to the level determined above.

Set audio gateway channel output levels to -10. Make a test announcement to each individual audio gateway, office and common area. Adjust audio gateway channel output volumes as required (self-amplified speakers and amplifiers connected to each audio gateway output should all initially be set to the same level (1/2 volume is typical).

Connect a music source or other repeating audio source such that the audio is broadcast across all speakers and horns at a level representative of a typical voice page[1], walk through the facility and verify that each speaker is receiving audio.

Make area specific volume adjustments as necessary.

If call buttons are installed, verify the operation of each VoIP Talkback speaker by pressing its associated call button and conversing. Make area specific volume adjustments as required.

If call buttons are not installed, verify the operation of each VoIP Talkback speaker by calling each speaker and conversing. Make speaker specific volume adjustments as required.

Involve the owner in the final testing to facilitate acceptance.

[1] Measure the level of a sample voice page with a sound level meter and then adjust the music or other source to a comparable level.

Product Specific Troubles/Tips

VE6023 Telephone Paging Server

If IP phones respond to the VE6023 group announcement by turning on, but do not broadcast audio, then the routers between the phones and the Valcom VLAN may not be properly configured to allow the multicast traffic to pass. To prove this, double click the phone in the VE6023 and uncheck the "Use Multicast" box. If the phone receives audio with this box unchecked, but not when it is checked, then multicast setup in the routers is likely the issue. It's important that the device sending the initial group audio to the VE6023 be on the same properly configured VLAN as the VE6023.

If only one simultaneous phone announcement is possible, add more multicast addresses in the VE6023 (one per simultaneous announcement).

If any of the IP phones don't appear in VE6023 scan, and the VE6023 is routable to the phones, then try reinitiating the phones by powering them down momentarily. The phones typically read a configuration file on boot up, part of which defines communications with the VE6023. If the problem persists, then the phones may have been set up with an improperly crafted configuration file. Contact us for additional assistance.

If phones on subnets foreign to the VE6023's subnet do not appear in the VE6023, and all phones have a properly configured configuration file, then the network routing should be evaluated. Also, if security controls (such as access control lists) have been established for the IP phones to communicate with the telephone system, they may not be able to communicate with the VE6023 telephone paging server.

You can test the routing by configuring your laptop NIC to use the same IP address as one of the phones in question and replacing the phone (use the same switch port) with your laptop. Try to ping, and or browse into, the VE6023. If unsuccessful then the switch port is not routable to the VE6023. This may not work if switch ports are locked down by MAC addresses or other security mechanisms. Refer to the *Testing Routing* section of this document.

Speaker Audio out of synch with Telephone Audio

The VE6023 listens to the same audio or text pages that is sent to Valcom IP speakers. The phone audio paging is slightly delayed due to the time required to contact the phones and initiate the paging function. Consequently, if there are IP speakers in the same listening area as phones, the audio may be played by the speakers before the phones. The Group Attachments function can solve this synchronization problem by coordinating the audio playback to both phones and speakers.

To use this feature, create two Valcom paging groups, one that contains only phones and one that contains only Valcom IP speakers. The audio page is then sent to the group containing only phones. The other group (containing only speakers) is then "attached" to the first group. The VE6023 will coordinate the audio delivery to make sure the audio streams are playing simultaneously. Refer to the VE6023 Manual for details.

In addition, Valcom Telephone paging servers feature a log file in their browser interface. This log file provides real-time information about server input and output activity.

When editing phones, multiple phones may be selected for common edits by clicking them while holding the Shift key and then double clicking to invoke the common change.

Note that depending upon the application, the VE6023 Telephone Paging Server and the VEUTM Unicast to Multicast converter may or may not play well with each other. If the VEUTM is expected to send audio to IP phone speakers, it will not work. The VE6023 changes the port/multicast address pairs for each phone announcement. The VEUTM uses consistently matched port/multicast addresses.

If the VEUTM will only be used to send audio to Valcom IP endpoints, then there is no issue.

Cisco Specific Notes:

If Cisco phones do not have the "Web access" setting enabled (on the Device page for each phone) you may experience http 401 errors.

In the CCUM Application User settings, make sure that all the phones you are paging to are included in the CTI Controlled Devices list.

Class Connection ES

Most Class Connection ES systems have more than one station card, more than one power supply and many speaker cable runs. This is the ideal situation for process of elimination troubleshooting. If a trouble is isolated to one station card, or one cable run, then swapping known working and trouble cards and circuits can quickly identify the cause of the trouble. For example:

If one station card with the address switch set to 2 is working properly, and a second identical station card with the address switch set to 3 is not, physically swapping the card locations can quickly identify if the trouble is associated with the card's position. If it is, then swap the Amphenols between cards to determine if the field wiring is causing the trouble. Be careful to make one change at a time so that you can identify which change corrects the trouble. Swap card positions, then Amphenols (speaker/call switch wiring), then card addresses (cycle CPU power after card address changes).

Example:

The station card with address 5 is in the 7th card cage slot and is not working properly. A station card with address 7 is in the 9th card cage slot and is working properly. The station card with address 5 is moved in the 9th card cage slot and the station card with address 7 is moved to the 7th card cage slot. If the station card with address 5 now works properly and the trouble has moved to the station card with address 7, then the trouble is related to the card mounting position and may be a power issue. If the trouble follows the card, then swap address settings to determine if the card address is the trouble. If so, the ribbon cable connection may be at fault.

The red LED on station cards indicates a trouble. Either the card is not communicating with the CPU or the fuse has opened. If the fuse has opened, a load issue in the field wiring is the likely suspect. See [Assessing Field Wiring](#).

Troubled speaker/call switch circuits may also be swapped with similar working speaker/call switch circuits to determine if the issue is related to the station card output circuit or the speaker/call switch circuit itself. It is advisable to assess the field wiring before moving suspect speaker/call switch circuits to known working station card outputs.

The system only has one CPU card, unless you have a spare, then swapping it with another is not a troubleshooting option. There are LED indicators on the CPU to indicate status. The most basic are the single flashing "heartbeat" LED on the lower front of the card and the "power rail" LEDs above the power connector. All should be illuminated.

Network connectivity

Network connectivity issues may be related to network routing or setup. Use a peer-to-peer setup from a PC directly to the CPU (use a crossover cable as required) to determine if the trouble is network setting related. If you can connect peer-to-peer then the CPU NIC (network interface card) is operating properly. Note that, for this test, the PC NIC must be set to a static address on the same subnet as the CPU.

Note that current CPU versions feature a 100 Mbps full duplex NIC, previous units had a 10 Mbps ½ duplex NIC. Appropriate switch port setting may be required.

Uploading WAV files

WAV files must be in an 8-bit ulaw format. Valcom offers a [WAV File Format Converter](#) software utility free of charge.

Some PC USB ports have difficulty uploading WAV files to Class Connection ES CPUs. Often, routing the USB connection through a passive USB hub will overcome USB port issues. Valcom can provide one free of charge is needed.

Appendix B of the Manual clearly outlines the upload procedure and a video may be found [here.](#)

VE8025/VE8045 Gateways

If 8-port gateways are going into a constant reboot state where both Ethernet LEDs flash in sequence, then the issue may be related to PoE switch firmware.

Not all 8-port gateways on a switch will necessarily experience the issue. Switching ports with a working gateway, the problem will follow the gateway. The Cisco switch logs may not show any data or PoE errors. The 8-port gateways will appear to be out of box failures but will function properly when moved to an unmanaged PoE test switch.

For example, we've experienced this with Cisco WS-C2960X-48FPD-L switches running 15.0(2a)EX5 firmware. Upgrading the firmware to version 15.2(4)E6 solved the trouble.

As soon as the firmware update was completed all units functioned properly.

VERCA Remote Card Adapters

When VERCA cards are used in a system, one is set as a "master" and is directly connected to the system CPU card via the system 40-pin ribbon cable. Up to 15 slave RCA Card Adapters are located remotely and extend the 40-pin ribbon cable via a [properly configured network connection](#). The master RCA card communicates with the slaves via multicast. If multicast is being received, each slave RCA card should have its MCR LED (D24) flashing rapidly. The main RCA card sends multicast, so it's MCR LED should be always be inactive.

LED	Board Address
D22	0
D21	1
D20	2
D19	3
D18	4
D17	5
D8	6
D7	7
D6	8
D5	9
D31	A
D30	B
D29	C
D28	D
D27	E

Other LEDs	
D24	MultiCast Reception (OK when Blinking)
D26	System Heartbeat
D23	Reset Active

There are LEDs on each RCA card (refer to the chart to the right). The master RCA should have an illuminated LED for each station card address associated with a slave RCA. The master RCA will not have an illuminated LED for the station cards to which it is directly connected.

The slave RCAs should have an illuminated LED for each station card address to which it is directly connected.

User's may also browse into the master RCA card and view active slave RCAs and their associated card addresses under the System Status menu.

ECH : Remote Card Adapter - Main

System Status

Key: blank = inactive, * = active, **D** = duplicate address

Id	Board Name	MAC Address	IP Address	Active	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E
01	Floor-2	00:D0:5F:00:6C:08	192.168.20.32	*	*														
02	Floor-3	00:D0:5F:00:6C:03	192.168.20.33	*			*	*											
03	Floor-4	00:D0:5F:00:6C:05	192.168.20.34	*					*										

(Navigation: Home, Network, Boards, Mappings, Copy Mappings, Password, Firmware, System Status, Reboot System, Logout)

Refresh

If the master RCA is not communicating with the slave RCAs, then the users will not be able to make announcements to the remote cards. Lack of multicast communications is the most common cause of this failure.

Improperly configured IGMP may result in the slave RCA cards being periodically removed from the IGMP group. If power cycling the slave RCAs temporarily restores operation, then they are rejoining the IGMP group at bootup, but the group membership is not being properly maintained by the network.

Disclaimer

Note that any applicable standards by official regulatory agencies or ANSI/*TIA*/EIA/IEEE should always be observed. In the case of conflicting information, these standards shall prevail.

The suggestions provided may or may not be suitable for your intended application. Please consider this information carefully before incorporating it into your system design. Valcom disclaims any responsibility for accuracy or completeness.

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We'd like to hear from you!

Valcom, Incorporated

5614 Hollins Rd

Roanoke, VA 24019 U.S.A.

1-540-563-2000

Regular Business Hours are Monday – Friday

8:30 a.m. to 7:30 p.m. EST

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