Tactical VoIP Intercom and Conferencing Systems

April 16, 2015
1. INTRODUCTION

This White Paper details how BridgeWay gateways and audio servers may be used to implement multi-node VoIP intercom systems with unmatched performance, scalability and versatility. BridgeWay technology solves the major VoIP communication problems of latency and QoS, while retaining the important advantages VoIP offers over traditional intercom system implementations. A few of these advantages are:

- Wide area networking
- SIP interoperability
- Programmable mission configurations
- Unlimited scalability
- Ease of installation
- Available fault-tolerant system configurations

BridgeWay VoIP networked intercoms and conferences support multiple simultaneous user talk groups. Talk group participants may include radios, telephone lines, SATCOM terminals, legacy intercom systems and networked VoIP devices, such as PC operator consoles, wireless PDAs, IP PBXs and SIP phones. Conferences may be assigned an extension number and are dial accessible by conference participants. Preset conferences may be created by the BridgeWay System Administrator and ad hoc conferences may be created dynamically on BridgeWay operator consoles.

2. NETWORK CONFERENCES

The BridgeWay system administrator may assign an extension number and a name to a Network Conference (also known as a “talk group”), allowing telephony and external VoIP devices to dial access the conference. Network Conferences may be accessed by an operator console Client GUI at the discretion of the BridgeWay system administrator. Client profiles are used to configure which Network Conferences, if any, are available to each client.

A powerful feature of BridgeWay Network Conferences is that they may be assigned an IP Address and IP Port to allow them to link to Network Conferences on other BridgeWay devices. This permits the creation of large, global conferences that span across multiple BridgeWay systems. Network Conferences may be configured to use Unicast or Multicast IP communications.

3. INTERCOMS

Intercoms are preset conferences local to their host BridgeWay that appear as intercom selection keys on BridgeWay operator consoles and intercom terminals. BridgeWay intercoms are full duplex and may be configured to limit the maximum number of users who may speak simultaneously, but otherwise operate as a traditional “Hoot-n-Holler” intercom.

Network Conferences and Intercoms differ, in that Intercoms are not assigned an extension number, so they are not callable by any telephony or external VoIP device, as are conferences. If physical assets (telephony users or radios) or VoIP (SIP or H.323) users need to participate, or an intercom group must span across multiple BridgeWay units, then a BridgeWay Network Conference can be used to support the intercom function.

Client profiles are used by the System Administrator to configure which intercoms, if any, are available to each client. Intercoms may also be assigned names at the discretion of the system administrator. Intercom participants may also be identified on the intercom client GUI screen.
The BridgeWay Client GUI includes control over intercom left-right audio selection, incoming volume level adjustment and hot mic or PTT (push-to-talk) controlled mic. Left-right audio selection allows the console operator to monitor a selected mix of audio channels (for example, radios) on one headset earpiece, while audio for the selected intercom channel appears on the other earpiece. While intercom communications are full duplex, users may elect to mute their microphone or utilize PTT control of their mic. BridgeWay includes features to allow SIP and analog phone users to generate a PTT control signal.

4. **Unicast Intercom Configurations**

Figure 1 demonstrates a simple two node BridgeWay Intercom. Unicast works well for this application, since it requires only a point-to-point connection. Unicast has the advantage of simplifying the configuration of network routers.

Clients located at both nodes may use the network conference feature to communicate with each other. More clients may be added on either end and, in addition, telecom and VoIP users may dial into the conference channel if an extension has been assigned for this purpose.

Telecom and VoIP users may be configured so that they are only talking on the intercom line when they press a pre-assigned DTMF key, and a second depression of that key, or a different assigned key, will return them to the listen only mode.

---

**Figure 1 – Two Node Unicast BridgeWay Intercom**
When the Tactical Intercom extends to 3 or more nodes, the shortcomings of Unicast become obvious, as shown in Figure 2.

In Figure 2, two intercoms have been established using the Network Conference mechanism. BridgeWay 1 has a conference to each of the other two nodes. In order to allow nodes two and three to communicate with each other, however, a patch must be created on node one that links their conferences together. This not only uses more resources on the system, it adds latency to signal pathway and creates a single point of failure, where if node one is taken off-line, nodes two and three are no longer connected to one another.

In short, the signal latency and network vulnerability attributes of Unicast is unsatisfactory for implementing large, multi-node, tactical intercoms.
5. MULTICAST INTERCOM CONFIGURATIONS

For this discussion, intercoms are implemented using the BridgeWay conference feature. Multicast holds numerous advantages for large, multi-node BridgeWay Tactical Intercom implementations. Figure 3 demonstrates a four node network using Multicast. While this network example consists of four nodes, the number of nodes that may be connected is unlimited when the nodes are connected by IP with Multicast capability.

In each node of this example, operator terminals and radios have been configured to participate in Conference 1, or Conference 2, or both. There may also be telephony or VoIP assets at any node that have dial access to either of the conference groups. In the Multicast VoIP configuration, an intercom central processor unit is not necessary, thereby eliminating single points of failure that may disrupt communications.

Multicast networks support multiple conferences/talk groups and, while the diagram illustrates two talk groups, clients on each BridgeWay unit may subscribe to up to twenty different multicast talk groups. Any client on a BridgeWay system may be a member of any Multicast intercom group or groups that the node has access to, based on their profile configuration. The maximum size of a Multicast intercom group is limited only by the network capabilities and the designated “life” of the Multicast packets, which controls how far they will travel from the source, before the network no longer forwards them.

**Figure 3 – MULTICAST BRIDGEWAY INTERCOM**
In a Multicast environment, any node may join an intercom group by alerting the network that they wish to do so. Their local router then alerts other routers in the network that it has at least one group member for the specified group. When any BridgeWay node sends packets to a Multicast group address, they are retransmitted to all members of that group. A member may withdraw from the group at any time without affecting others in the group, or causing loss of service.

6. DESIGNING SUPERIOR PERFORMANCE INTERCOM CONFIGURATIONS

This section provides an overview of how to configure a VoIP intercom system offering superior performance parameters. Superior performance may be defined as exhibiting one or more of the following characteristics:

- Low latency, high quality, audio
- Flexible integration with associated C4I equipment
- Unlimited scalability
- No central controller and no single points of failure
- Simplified cabling and installation
- Reduced SWaP over legacy implementations
- Wide area networking
- Auto failover/recovery from a LAN or system component failure

6.1 OVERCOMING THE LIMITATIONS OF PC CONSOLES IN INTERCOM SYSTEMS

PCs are often used for operator position VoIP communications, with reliance on the PC audio subsystem for the user audio interface. The PC audio subsystem is a low priority task that is prone to delays and interruptions, causing latency and QoS degradation. The added latency additionally creates significant problems when operators are in close proximity, creating echo, and generating unacceptable delays when multiple sites are networked over satellite links.

The following section details how the Tactical Communication’s VHI VoIP Audio Gateway may be used to resolve the problems of latency and QoS, while offering the advantages of unlimited scalability, enhanced collaborative communications, no single points of failure, reduction in equipment compliment and ease of installation and use.

In Figure 4, operator positions are equipped with the Tactical Communications VHI VoIP Audio Gateway (VHI). Each VHI supports 2 operators with audio conferencing, SIP communications, call management, custom configured operator and console position profiles, and a client GUI server. The VHI supports military and aviation headsets, including ANR types, with PTT or “hot mic” features for each operator position.

When operators are in close proximity, VoIP delays create an echo, which can disrupt and confuse communications. The VHI transports audio from the network directly to the user headset, providing significant reductions in latency, compared to delays inherent in the audio system of a typical PC. The embedded VHI conference bridge provides audio monitoring/mixing and conferencing of multicast audio streams, eliminating the necessity for a separate networked conferencing server and its associated latency contribution.

By virtue of its embedded conferencing bridge, each VHI functions as a standalone audio server, eliminating the necessity for a central control unit, permitting intercom system scalability to an unlimited number of participants. As an added benefit, the failure of any VHI will not affect the operation of other VHI units on the network.

Use of the VHI also allows users to run mapping, video, or other applications on the Communications Control PC without risking audio signal interruption. The VHI system embedded client server eliminates the necessity for dedicating a VoIP communications console to each operator, allowing a pop-up GUI to run on existing console PC’s and workstations.
Operators may login to any VHI on the network from their PC console. Since each VHI is equipped with two independent operator audio ports, operators in close proximity to each other may plug their headset into another operator's VHI unit in the event of a failure of their VHI unit. By virtue of this feature, the VHI serves as a building block for fault-tolerant configurations exhibiting no single points of failure. Auto failover operation may be achieved by adding the Tactical Communications Audio Mixer Unit adjunct to the VHI. The Audio Mixer Unit automatically performs a cross connect, without operator intervention, in the event of a VHI failure.

An important availability feature offered by the VHI allows the operator to continue radio and intercom audio communications in progress uninterrupted, even in the event of a failure in their console PC used to control the VHI. In the event of a failure of an operator PC console, a spare console on the network may be used to login to the VHI.

**Figure 4 – Tactical Intercom with VHI Audio Gateways**
6.2 Satellite Networking Considerations

Intercom systems networked via satellite must be designed to handle the unique characteristics of SATCOM networks, including link latency, high packet loss and limited bandwidth. TCP/IP is unusable in a SATCOM environment, as network latency can exceed 250 msec. and retransmission of lost packets is not feasible. Instead, UDP/IP, coupled with specialized signaling protocols, must be used.

*The specialized signaling protocols ensure reliable delivery of command and control data, such as PTT on/off signaling, when using the UDP/IP protocol in a high packet loss environment.*

A robust voice compression algorithm that can withstand high packet loss and still deliver high quality audio is typically necessary to operate within SATCOM link bandwidth limitations. Silence detection algorithms may be used to minimize packet transmission and bandwidth utilization when audio is not present.

*Tactical Communications has successfully employed G.729, G.729D and MELPe in SATCOM linked environments.*

Early on, SIP was investigated by Tactical Communications as a potential call control protocol. The limited bandwidth of the satellite link, combined with limitations imposed by the SIP protocol, precluded the use of SIP. In its place, Tactical Communications developed a proprietary communications protocol for satellite networks, field proven in multiple UAS, ground-air and satellite networked multi-site networked programs.

*Tactical Communications protocol improves SATCOM bandwidth efficiency over SIP, especially in cases when multiple individual SIP calls are placed to the same airborne destination.*

The contribution of VoIP ground systems to total end-to-end communications latency must be minimized to compensate for the unavoidable satellite link latency.

*Tactical Communications VHI system has proven valuable in achieving absolute minimal VoIP latency in ground communication systems used to support SATCOM linked networks.*

6.3 Fully Redundant Configuration Tactical Intercoms

In the multicast configuration, BridgeWay equipment failures at any one site will not affect ongoing communications at any of the other sites. If failure *within a site* is not an option, BridgeWay Gateways and VHI Audio Gateway systems may be configured for redundant LAN and gateway operation to eliminate all single points of failure.

Refer to Figure 5. In this configuration, a pair of cross-connected BridgeWay systems, multiple VHI systems paired with Audio Mixers, and dual LANs provide a fully redundant system with no single points of failure. In this example, the BridgeWay Gateway supports a variety of telecom line types and multiple radio ports.

The operator command and control computer GUI may be configured to sense component failures and seamlessly switch to a backup system, LAN or VHI audio server, allowing uninterrupted communications. For further details, refer to Tactical Communications Application Note AN-1814, “A Fault-Tolerant VoIP Tactical Communication System”.
6.4 Tactical Vehicle Intercoms

A VoIP tactical vehicle intercom system with superior performance may be configured using Tactical Communications intercom terminals. Tactical Communications offers crew terminals with rotary switch selection of the intercom channel, as well as a mode switch to select between several pre-programmed configurations. Advantages of the VoIP-based intercom system are:

- The TCC-2 VoIP intercom terminal is mission configurable over the network
- Simplified installation – the PoE version plugs into a single LAN cable
- Supports redundant network connections
- Scales to very large number of users
- No central processor unit to create a single point of failure
- Ability to monitor multiple intercom channels
- Information display option shows intercom participants, speaker ID and activity on monitored intercom channels
- Crew members may have unique configurations selectable via the mode switch and auto loaded upon login
- Crew members may roam and their custom configuration will follow
- Encryption option
- Active Noise Reduction (ANR) option
7. **BridgeWay Systems Communications Console Options**

Tactical Communications offers a variety of intercom terminal devices to enhance communications capabilities for tactical communications vehicles, shelters and on the move combatants.

7.1 **TCC-2 VoIP Operator Console**

The TCC-2 is a self-contained, MIL-STD VoIP operator console designed to operate in shelter and mobile vehicle environments. TCC-2 supports military or aviation ANR headsets and is powered from platform DC power or PoE. TCC-2 runs a fully featured BridgeWay operator console GUI via its touchscreen or gloved hands capable interface.

![Figure 6 - TCC-2 Operator Console](image1)

7.2 **VHI Audio Gateway System**

The VHI is a MIL-STD VoIP communications audio server and conferencing system designed to support operators performing data intensive command and control operations. The VHI is controlled by a thin client running on the operator computer.

![Figure 7 - VHI-01M (MIL-STD version)](image2)
7.3 MOC-1 On-the-Move Operator Console

The MOC-1 is a MIL-STD battery operated PDA operator console designed for on-the-move applications. The MOC-1 runs a subset of the BridgeWay operator console GUI via its touchscreen or cursor driven interface. The MOC-1 Wi-Fi connection automatically links to a supporting BridgeWay system upon arriving at the site or platform. The MOC-1 will be loaded with its appropriate mission profile and client configuration upon link-up.

![Figure 8 - MOC-1 Operator Console](image)

7.4 CS200 VoIP Crew Station

The CS200 is a MIL-STD single or dual operator intercom crew station, featuring programmable front panel rotary switches with optional keyboard and display. The CS200 supports single or dual LANs, one or two crew positions and is powered by 12–48V DC platform power or PoE (Power over Ethernet) for simplified installation and plug and play operation. Features include selectable hot mic or PTT controlled audio; binaural left-right, or monaural, headset audio; built-in speaker monitor amplifier and optional active ambient noise cancellation.

Various software programmable switch configurations are available for the CS200 series products. A single position crew terminal typically is configured with an intercom channel selection switch, a mode switch to select among several pre-programmed user or terminal specific profiles, a headset volume control and a monitor speaker volume control. A dual position crew terminal typically is configured with dual intercom channel selection switches and dual headset volume controls.

![Figure 9 - CS200 Intercom Crew Station](image)
8. VoIP Console GUI Intercom Screens

Tactical Communications offers a configurable GUI interface for operator controlled intercom systems. Figure 10 illustrates a typical combined softphone and intercom GUI.

The softphone can support up to four calls/intercom sessions. A volume slider is provided to control the receive volume, along with a mute button. A list of active users is displayed in the Users window. Selecting an active user and pressing the off-hook button will place an intercom call to the selected user. Once an intercom session is established with another user, it will be added to the next available line. An intercom call is accepted by pressing the off-hook button. An intercom call may be terminated by selecting the call and pressing the on-hook button.

An audio and visual indication is provided for incoming calls. The calling party will be added to the User window, as shown in Figure 10. The call is accepted by pressing the off-hook button. Once the call is accepted, it will be added to the next available line and the name is removed from the User window.

An outgoing call can be made by either, pressing the off-hook button and dialing the destination number, or by first dialing the destination number and then pressing the off-hook button. An outgoing call will be added to the User's window as shown above until the other party answers. Once the other party answers, it will be added to the next available line and the name is removed from the User window. An active call can be terminated by selecting the call and then pressing the on-hook button.

A call can be put on-hold by selecting the call and then pressing the hold button. Toggling the hold button again will remove the call from hold. A call can be muted by selecting the call and then pressing the mute button. Toggling the mute button again will un-mute the call. Call transfer can be accomplished by selecting an active call and then pressing the transfer button. This will automatically place the call on-hold, provide a secondary dial tone and allow for dialing another party. Once the other party answers the call, the call will transfer by pressing the transfer button again. The transfer will remove the call from the active line list.

Conferencing can be accomplished by selecting an active call and then pressing the conference button. A secondary dial tone is generated and dialing to another party is allowed. Up to three additional members may be added to the conference in this manner. Pressing the conference button twice without any further dialing will conference all the parties.

The Tactical Communications GUI includes a language translation module, programmable by the system integrator, to display foreign language text on the user screens. The Tactical Communications GUI is written in JAVA, with Android and Microsoft .NET versions scheduled for delivery in 2015.

Tactical Communications will also license a custom GUI development package, including documentation, technical support and example GUI software, to those customers that prefer to develop their own GUI for BridgeWay products.
Figure 10 – Example Softphone/Intercom GUI Screen