IPS Radio Interoperability Gateways

Overview

The system is a real time VoIP communications platform. It features “one touch” operation that allows users to make instant calls and start communicating right away. It is comprised of a suite of products that are versatile enough for scaling out on LAN, WAN, Internet and over Satellite. These products can support direct communications or be interfaced to external devices to provide interoperability.

Utilizing a user friendly approach makes mastering the powerful capabilities intuitive. Behind the scenes is a feature rich set of tools that allows the system to be configured and administered securely. The audio quality is superior to that of typical radio and telephony.

The system uses a building block approach with peer-to-peer architecture with no central servers or single points of failure required for the system to operate. The maximum capacity for the system is 64,000 devices to support over two million users. This allows flexible network design to optimize functionality and reliability.

User Interfaces

Users access the system by a variety Virtual Panels (PC users) or Hardware Panels (non-PC users). These panels can be provided to individual users or dispatch personnel anywhere on the network. Panels for dispatch locations do not require any direct connection to existing dispatch consoles.

Users can participate in or monitor multiple voice conversations at once. They are not limited to participating in just a single call at a time. All users enjoy a similar experience with “one touch” control. Making calls, participating in talk groups or arranging calls for external devices can be performed with the simple push of a button. All users get real time color coded status indications of active connections. All users get easy to access controls to customize or manage their areas of responsibility.

Users can choose from a wide range of panels:
Virtual Panels are PC applications that provide a GUI for users. These allow users to communicate with each other or to provide a console for operators to manage talk groups with external devices. These can be configured by the Administrator or customized by the user as appropriate for a wide range of functionality and appearance. Virtual panels can run on Microsoft Windows 98SE, ME, 2000, XP Pro; a Pentium II-450 minimum is recommended.

Hardware Panels do not require a PC GUI to operate. They are dedicated devices available in a range of offerings with choice of form factor and size. Stylish desktop panels as well as console panels are available. All panels have integral microphone and speaker, headset connector and display.

External Interfaces

LMR Interfaces

The system can integrate with all types of LMR devices including:

- Conventional Analog
- Conventional Digital
- Analog Trunked
- Digital Trunked
- Hybrid/Mixed Mode Analog/Digital Conventional
- Hybrid/Mixed Mode Analog/Digital Trunked

It uses analog audio input and output to support LMR voice signals. It can use PTT, COR, E&M or VOX to support LMR signaling. The system has an assortment of adjustable parameters to facilitate integrating each LMR device into the system.

Radio Interoperability Board

The Radio Interoperability Board (RIB) is an 8-channel radio expansion board for Black Box 2.5 that features:

- Interconnection of radio PTT, COR and audio directly on each RJ45 connector.
- Audio delay on all inputs and outputs (25-1000ms)
- Balanced and unbalanced audio
- Selectable -20dB output pad built in to drive the mic inputs of radios directly.
- Selectable +20dB input boost

PSTN

The Dual Channel Telephone Interface is a 1RU x 19” frame that houses 2 channels of PSTN (POTS) interface. These can be connected to PBXs or POTS lines and can be directly connected to a Black Box 2.5 unit fitted with an AEB to provide PSTN interface to the Mercury system.

Cell phones & Walkie Talkie Phones

Cellular handsets and cellular base stations connected to the system through a cradle or special cable. This enables direct connection to a port where the device is to interoperate with other devices on the system. Additionally, third-party interfaces are available that can adapt cellular handsets to PSTN and then connect to the system through PSTN interface.

Hard-wire Interface

The system has analog audio inputs and outputs readily available on the Audio Expansion Board (AEB) for universal interconnection to external devices. It also supports a wide range of configurable parameters to integrate each device into the Mercury system.

Satellite Phones

Satellite phones interface to the system by special cable. This enables direct connection to a port configured to interoperate with other devices on the system.

E&M

The system interface is a 1RU x 19" frame that houses 16 channels of E&M interface. It has selectable E&M signaling types I, II, III, IV and V and 600ohm transformer coupled balanced audio I/O. Then connected to radio or telephone E&M circuits and then directly connected to a Black Box 2.5 unit equipped with a 700-12-00 General Purpose Interface (GPI) module and an Audio Expansion Board (AEB) to provide E&M interface.

VOX

VOX is available on any device connected to the input of the system, and configured on any input by the Configuration and Editing software. This allows for the adjustment of multiple parameters including VOX threshold, delay hold off, sustain, and triggers for GPI events. Adjustable parameters include VOX threshold, attack and release.

PTT

Push-to-talk (PTT) is available on each channel of the Black Box 2.5 Radio Interface Board and on each channel of the GPI module. PTT is also available as a serial port connection on any PC running a Virtual Panel application. This allows any device to communicate a PTT command to the Mercury system or conversely, for the system to communicate a PTT command to the external device. Interfaces can be configured variously for opto-isolator, TTL, serial and open collector.

COR

Carrier Operated Relay (COR) is available on each channel of the Black Box 2.5 Radio Interface Board and on each channel of the GPI module. This allows any device to communicate a COR command to the system. Interfaces can be configured variously for opto-isolator or TTL. Adjustable parameters include COR attack and release.

Types of Calls

Typical call types include:

- Individual calls – direct point-to-point calls between users. Multiple calls can be made at a time.

- Group calls – from one user to many users. This allows “one touch” action to reach a predefined set of users. This can be extended to make announcements and “all calls”. Multiple group calls can be made at a time.
- Talk Groups – conferences that can be joined by multiple users. Panel users can join themselves into talk groups. External devices can be joined into talk groups by dispatch operators or other users that have the necessary privileges. The system configuration by the Administrator supports virtually an unlimited number of talk groups. These can support a wide range of governance rules and protocols. Users’ priorities can be developed by arranging users into suitable talk groups, by allowing users to participate in multiple talk groups simultaneously and by assigning user levels of priority. The system can support FIFO for radio users of similar priority in talk groups. Call priority permits a call originating from a dispatch operator to interrupt a call by a normal member of the same talk group, overriding the call so that other members of the talk group only hear the dispatcher.

Routes

The system supports a wide range of static and dynamic routing types over the IP network. The routes can be defined by the Administrator in the Configuration and editing software and deployed as permanent routes or exposed to operator(s) for dynamic routes. Disparate external devices can be integrated into the Mercury system with suitable interfaces and appropriate Configuration and Editing parameters.

Simulcast LMR transmissions

These are usually associated with multi-site cross-band repeater applications. Setting up these types of applications is similar to setting up talk groups in general. There is typically one speaker at a time to the talk group and many listeners. There are two special concerns with simulcast LMR transmissions:

- A portable radio listener who is in an RF reception area that is overlapped by more than one simulcast repeater transmitter on the same frequency where the RF signals are within 6db of each other can demodulate both signals simultaneously. In this case the signals need to be synchronized within a fine tolerance in order to prevent an echo from being perceived by the listener. The Radio Interoperability Board has independent audio delay on each input and output. The audio delay range is 25-1000ms (can be extended to 2000ms if required). Resolution is down to 1ms increments.

- Full duplex operation – Each repeater may be either a listening member of the talk group or a speaking and listening member of the talk group. Therefore, the system must be capable of dynamically reassigning each repeater’s input and output to correspond to different portable radios becoming speaking members to the talk group across a multi-site cross-band repeater system. This entails being able to divorce a particular repeater’s speak circuit from the listen circuit in a manner that automatically responds to the changing conditions.

- Architecture

The system has a wide range of deployable products for LAN, WAN and Internet. These include the Black Box 2.5, PCI cards and USB Adapters. Each product can host user interfaces and connect to external devices. These are all similar in design and differ mainly in their capacity. The Black Box 2.5 can host many user interfaces and external devices. It is suitable in areas that support multiple users. The PCI cards and USB adapters can host two user interfaces or external devices. These are appropriate in areas that support one or two users.
The Black Box 2.5 is at the heart of the system for providing radio interoperability solutions. It is a 2U x 19" System Interface Frame that provides an interoperable GATEWAY to the IP network. Each Black Box 2.5 unit is equipped for direct connection to the IP network with a single 10/100 Base-T Ethernet connector (RJ45). Any standard IP connection to the IP network can be utilized – no special routers or switches are required.

Each Black Box 2.5 unit is equipped with a 66 x 66 non-blocking TDM matrix. This provides mixing, switching and routing of a wide range of voice and audio signals. Each Black Box 2.5 unit accepts a variety of expansion boards that can be appropriately fitted to suit application requirements.

The Black Box 2.5 may be fitted with up to four optional expansion boards. The Audio Expansion Board (AEB) provides eight RJ45 connectors for eight channels of 4-wire audio or intercom panels. The Radio Interoperability Board (RIB) provides eight RJ45 connectors for direct connection to eight radios. Any mix of these expansion boards may be combined to support up to 32 channels of interface to external devices per Black Box 2.5 unit. As more channels are needed, simply add another Black Box 2.5 unit fitted with the appropriate expansion boards. In this manner the system can be scaled out with a building block approach.

The VoIP Interoperable System is comprised of one or more Black Box 2.5 units. These units provide the IP gateways for the required external devices (i.e. radios, telephones, intercom panels, user stations, etc) to be combined into talk groups across the IP network. This allows for static or dynamic assembly of any combination of real time interoperable voice or audio assets across the IP network. These assets can be managed by Virtual Panels (PC GUI control surface) or Hardware Panels (non-PC control surface) connected anywhere on the IP network. This allows for dispatch operators to control any range of assets. It also allows for individual users to participate in or to monitor voice conversations.

Administration

The Administrator(s) Configuration and Editing software manages the peer-to-peer devices and allows for secure centralized administration. Single or multiple administrators can make configuration changes from anywhere on the IP network, and transmit them to all system devices on the IP network from a central point or from within zones. This permits flexible administration design for a range of security, reliability and resource criteria.

The PCs operate on or off line from the network and the application is a comprehensive tool kit that provides the Administrator with tools to manage IP network and the system parameters. It allows the creation and management of update files used to update and configure the devices on the network. It uses drag and drop as well as pull down techniques wherever possible to give the user a familiar Windows environment from which to operate. These files synchronize with a network database server or servers to transmit any changes to the system devices.

Some items that the administrator(s) can control are:

- Call types
- Talk groups
- How calls are exposed to users
- Static or dynamic routes
- System levels
- System bandwidth
- Coding profiles

VoIP

The system natively supports standards based VoIP technology. This permits seamless integration of IP voice communications over LAN, WAN, Internet and Satellite. It does not require any special routers, switches or gateways. It can be used with wireless networks. It is scalable over a distributed architecture and can traverse different bandwidth segments automatically. It supports NAT and can go through firewalls. Standards based encryption devices including VPN can be used transparently. The system provides robust performance on a shared common network with other services, including data and video.

IP over satellite is becoming popular and is growing rapidly. Innovative technologies and competitive pricing have made this transmission platform widely available for remote operations, back up operations as well as primary operations.

Security in the system is at the network level. The VoIP signals are part of the network traffic and can be viewed within the overall network security policies set by the network Administrator. Any standards based IP encryption device (VPN or otherwise) can be utilized.

External encryption devices like radios and phones can be used. All audio and voice signals that are interfaced to external devices are mixed, switched and routed at the audio base band level. The system does not encode or decode these signals. It passes them through transparently as they arrive.

The system allows the Administrator to customize the overall system bandwidth requirements as well as any local bandwidth requirements. The Administrator can choose from a variety of coding profiles (or create a custom coding profile), which allows

precise control over bandwidth and voice quality in a distributed network environment. The parameters include the CODEC (industry standard G.722, G.711, G.723, G.726 and G.729), the jitter buffer size and the amount of audio to stuff into packets. This provides an operating range from 10Kbps to 100Kbps that can be selected for IP payloads of each active talking channel. Silence suppression can be utilized to further reduce IP payload by not sending out empty packets when audio falls below a prescribed threshold.
Multicast can be utilized to better manage IP bandwidth in a distributed architecture. This allows large numbers of users to participate in talk groups with bandwidth only required for the speaking member(s).

Low latency is characteristic of this system compared to other platforms. Broadly speaking, latency falls into two categories:

- Propagation Delay – This is the latency that is caused at the IP transmission level. This will vary depending mainly on how many IP routers the signal traverses. Typically these latencies are within 20ms on LAN segments, 20 ~ 40ms on WAN segments and 20 ~ 80ms on VPN segments inside the continental US. Fixed orbit satellite hops are typically 600ms.

- LMR Delay –With trunked radio systems and repeaters it is usually necessary to add some delay to the audio signal that is being transmitted by those systems. Artificially induced latency prevents leading edges of voice messages from clipping on any input/output of the Radio Interface Board. The Administrator has adjustable parameters in the Configuration and Editing software.

Reliability

The system components are peer-to-peer devices deployed over a distributed architecture. Peer-to-peer operation of the system allows for centralized and decentralized operation. Each device is self-contained and has everything that it needs to operate independently of other devices. If a portion of the network or its devices is unavailable, the rest of the system operates without them. If different segments of the network are disconnected the devices within each segment of the network continue to operate. This can extend right down to a single device that has no network connectivity at all. The device will still operate normally for any user interfaces or external interfaces that are connected directly to it.

No single point of failure is fundamental to the system architecture. Peer-to-peer system devices do not depend on any central server or device for normal run time operation. There is no wide area controller in the system architecture.

All the system devices use network “keep alive” signals to insure the connections automatically restore and maintain after an interruption.

Logging

The system provides open architecture access for third-party recording and logging systems. Analog outputs can be made available anywhere on the network for universal interface to most external devices. This mix can then be routed to any recording device located anywhere on the network.

The system integrates seamlessly with the Stancil Multi Channel Voice Logging Recorder System. The Stancil system can receive Mercury IP multicasts anywhere on the IP network. Therefore, there is no hard wire requirement between the Mercury system and the Stancil system. Additionally, there is no hard wire requirement between the required external devices (i.e. radios, telephones, intercom panels, user stations, etc) and the Stancil system. The Mercury system can provide the appropriate source audio to the Stancil system anywhere on the IP network. This facilitates a flexible cost effective approach to voice logging requirements.

The Stancil system has as standard an 80G hard drive that can store 12,000 channels of recordings at 13.3GSM. The hard drive can be increased for additional storage. Recordings are automatically downloaded to DVD-RAM drives for long-term storage. The clients access the recordings from TEN-4 clients running on a standard Windows 2000/XP network machine. The network remote client attaches to the recording system and finds the recordings by the users' preferred search criteria. This is typically by channel and data/time. The result set is presented to the user in a graphical manner.

System Management

The Database Supervisor shows all connected devices and applications. It can be viewed from any properly configured PC on the network. Additional status and diagnostic information about individual devices, software applications and external equipment status information (e.g. PTT, COR, E&M states and other logic states) can be viewed in separate windows. All devices have log files that produce a text file of detailed operational conditions. All control panels support real time status of active connections. The Black Box 2.5 has a rear connector panel that can be connected to an enunciator panel or higher order Network Management System to indicate status and provide remote alarming.

All of these tools can be utilized by the Administrator or maintenance staff to isolate or troubleshoot problems. This can even be used to deduce the status of devices that are not directly managed.

The Trilogy product roadmap includes the development of a Connections Visibility and Performance Measurement (centralized status/failure monitoring capabilities) software upgrade scheduled for release in Q3 2005. This will further enhance the System Management capabilities.

SDK

The system has an open API at the sockets layer. An SDK is available for developers. This allows third-party software and higher order Communication Management Systems to automate the system from the IP network. In this way, the system can be utilized to manage the required external devices (i.e. radios, telephones, intercom panels, user stations, etc) and to provide the infrastructure for VoIP talk groups. This provides compatibility for advanced features like channelization and integrated scenario actuation. This provides a means for other applications to consume the functionality and offer an integrated GUI.

Standards

The system uses the following standards:
IP, TCP, UDP, IGMP, TFTP, RTP
802.1P & Q, Diffserv, IPV6

References

- ✓ Tactical Military Operations
- ✓ Mobile Command Vehicles
- ✓ Operations Centers
- ✓ International News Operations
- ✓ News networks
- ✓ Training and Simulation
- ✓ Teleports
- ✓ Electronic Courtrooms
- ✓ Biometric Access Panels
- ✓ Radiation Screening Portals
- ✓ Interactive Distance Learning

References are available upon request.