Battle Command

Battle Command Radio Net User Guide

Historical Software Corporation - www.historicalsoftware.com

1.0 Introduction

The Battle Command Radio Net (BCRN) programs permit voice communications between participants while they are using the Battle Command simulation. BCRN can also be used in conjunction with other simulations or games, or as a standalone program to allow secure encrypted Internet voice communication and meetings. AES 256-bit encryption is used. The BCRN system consists of two programs: the BCRN Server and the BCRN Client. The system allows participants to set up multiple voice communication channels for use during simulation. Participants on Side 1 and Side 2 are each allocated five communication channels. Transmissions on these channels cannot be monitored by the other side opponents, or by any simulation observers. A special "simulation-wide" channel can be monitored by both sides and by any participating observers. Observers are allocated their own dedicated channel (transmissions on this channel cannot be heard by Side 1 or Side 2. This combination of channels results in a total of twelve BCRN channels.

Participants choose what to name each channel, which channels to monitor, and what channel to transmit on at any given time. Before simulation use, users should decide what each of the allotted channels will be named and used for. For example, brigade command could be assigned Channel 1, 1st battalion Channel 2, 2nd battalion Channel 3, support artillery Channel 4, and logistics Channel 5. Staff running the BC server can communicate with all participants using the "Sim-Wide" Channel. Participants can also use this channel to communicate with the other side if necessary, or with the simulation controllers and observers.

To use the system, the BCRN Server can be run in parallel with the Battle Command Server, possibly on the same LAN or PC. Participants then register into the BCRN Server by pressing the "register" button in the Client so they will be included in the distribution of communication messages. Participants then select a transmit channel and press the talk button on the Client before speaking into the computer microphone. Anyone monitoring that specific channel on the system will be able to hear the transmission and respond.

2.0 Setup

The Radio Net system uses the UDP transmission protocol. As a result, it is possible packets can be dropped and depending on network congestion or slow connection speeds this may become more noticeable. Under normal conditions intelligibility is excellent as BCRN uses a transmission approach identical to the systems used in many high-quality commercial VoIP systems.

Setup Steps:

1. OPEN UDP PORTS. If the Server and/or Clients are run behind a router or firewall, port forwarding to the Server and/or Client computers must open up UDP access on ports 50,000 to 50,200 and forward packets to the computer running the Server and/or Client. The system makes use of a portion of these 200 ports for UDP packet transmission.

2. RUN THE SERVER. Identify the WAN (Internet) IP of the Server computer. If the Server is behind a router, check the router for the WAN IP address, which will not be the same as the computer's LAN IP address (for example: 192.168.0.101 is a LAN IP address and will not work over the Internet). If you wish for participants to hear their transmission echoed back to them (for troubleshooting or other purposes), select the "Echo Back" checkbox on the Server window.

3. RUN THE CLIENT. Insure at least one computer microphone and sound card are working properly before starting the Client. Each participant wishing to communicate must run a Client. After starting the Client, enter the Server LAN or WAN IP address and click Register as one of the following: Side 1, Side 2, or as an Observer. Choose this button depending on which group you are a member of. If the Client successfully registers, the light will turn green next to the register option.

4. COMMUNICATE. In the Client, enter what to name each channel if desired, select which channels to monitor, select a channel to transmit on, and press the "Press to Talk" button to begin communicating.

3.0 BCRN Server Window

Below is the Radio Net Server main window.

After participants register correctly, their respective WAN/LAN IP addresses will be displayed in the IP address list of the side they registered on. IP address 127.0.0.1 is the loop back address and represents the same computer.

Channel Activity: Light turns green if there is voice traffic on the respective channel.

Registered Users IP Addresses: Shows the IP address of registered users and user count under the side they have registered for.

Sound Quality: Choose the sound quality and compression for connected users. Both uncompressed and compressed formats are available. CBR stands for "constant bit rate", and VBR stands for "variable bit rate". The default value (Compressed, 32 kHz, 16 Bit, Mono, VBR) will give superb voice quality performance and extremely efficient compression (usually > 90%), however lower levels may provide smaller transmission bandwidth requirements (in terms of kB/sec). Higher voice quality (bandwidth) levels may not always work well depending on Internet connection speeds and number of connected users. After the first user connects, selection of sound quality becomes disabled in the Server.

Compression Quality: This slider will set the relative quality of compressed voice transmission formats. A value of 0 is low-quality, and 10 is the highest quality. VBR settings will normally produce better quality audio at the same compression quality slider setting as the corresponding CBR option. The following table shows the approximate transmission bandwidth requirements for various settings:

Compressed Format			Approximate Kilobytes per Second			Overall
Frequency	Bits	Data Rate	Quality=0	Quality=10	Uncompressed	Reproduction
8 kHz	16	CBR	1.22	3.72	16.00	Excellent
8 kHz	16	VBR	1.04	3.16	16.00	Raspy
16 kHz	16	CBR	2.11	5.94	32.00	Excellent
16 kHz	16	VBR	1.42	4.00	32.00	Good
32 kHz	16	CBR	2.03	6.14	64.00	Superb
32 kHz	16	VBR	1.66	5.01	64.00	Best/Superb

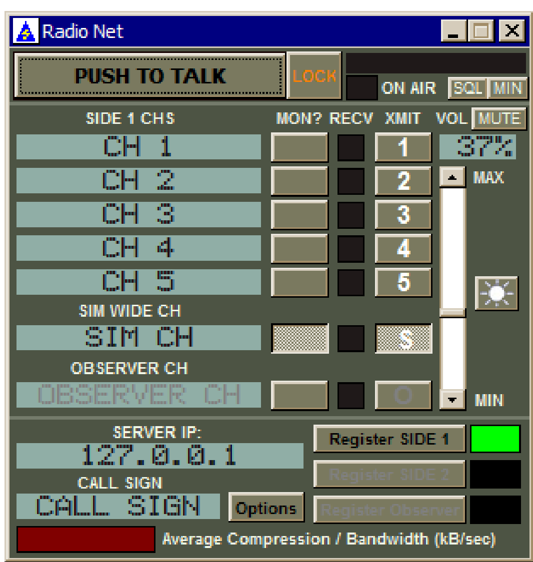
Last Data Sent: Shows the last UDP packet transmission time to pass through the server.

Require Encryption: Indicates if voice transmissions will be encrypted, which involves a small degree of additional CPU processing overhead. If selected, all users must input the system-wide password encryption key into their client to be able to decode transmissions. All users must be using the same key string value to be able to successfully encrypt and decrypt transmissions. Advanced Encryption Standard (AES) 256-bit Rijndael encryption is used.

Echo Back: Relays voice transmission back to the originator with a delay of a couple seconds. This can be useful for troubleshooting or to hear how your voice will sound to other participants.

4.0 BCRN Client Window

Below is the Client main window. Access to the Client on ports 50,000 to 50,200 for UDP transmission from the BCRN Server is required.



PUSH TO TALK: Press this button to transmit. Release the button when finished. There is a mic "key down" sound that is played locally and with the transmission, and a mic "key up" sound when the talk button is released. These sounds can help to identify the beginning and end of transmissions, and can optionally be turned off.

LOCK: Locks the mic key down and causes continuous transmission.

ON AIR: This light will turn red during transmission. If the light goes black, it means no transmission is occurring.

MIN: This option will allow you to cycle through main window sizes and shrink the main window so only the talk button is visible, expand it so that the channel information is also shown, and expand it further so that the compression status is shown in addition (default).

SQL: Activates "Squelch Mode" (voice activation of transmissions). Set the "Squelch Level" on the Options Window to set the sensitivity of when the microphone will begin broadcasting.

SERVER IP: Participants must manually enter the Server WAN/LAN IP address given to them before clicking a register button. If you are connecting to a Server being run on the same LAN, the LAN IP address of the Server computer is the correct input here. If the Server is being run over the Internet, the WAN IP address of the Server must be entered and the router at the location must have port forwarding set for UDP on ports 50,000 to 50,200 to the Server computer.

CALL SIGN: Allows the user to type in an assigned call sign they can use for reference.

OPTIONS BUTTON: Brings up the BCRN Client Options window.

REGISTER OBSERVER / SIDE 1 / SIDE 2: The participant selects the side they wish to register as and communicate with. It is important to select the correct side, registering for the incorrect side will allow one to eavesdrop on the other side's communication and is prohibited. Once correctly registered, the light next to the selected side will turn green. Further registration for other sides will be disabled.

NAME CHANNEL: There is a text box for naming the Simulation-Wide Channel, the Observer Channel, and each of the five Side Channels. You can assign names for each channel by typing it in the respective name text box. Naming of channels is not a requirement.

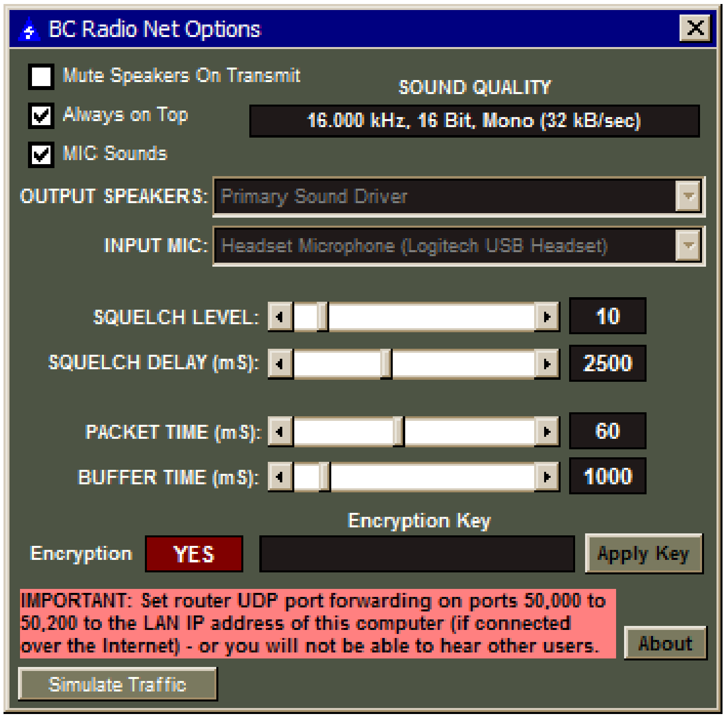
MON?: Use these buttons for selecting to monitor specific channels. If the selection is off, transmissions on the channel will not be audible.

RECV: This light will turn green during the time there are transmissions on the respective channel.

XMIT: Use this option button to select which channel to transmit on. Some channels (depending on the registration option) may be grayed out and unavailable. You can only transmit on one channel at a time. Ask other participants to tell you if your transmissions are too loud or too soft, and vary your distance to the microphone accordingly to achieve the best sound level. Generally, about 2-6 inches is best.

VOL: Use this slider bar to set the sound volume for incoming voice transmissions.

5.0 BCRN Client Options Window



Mute Speakers on Transmit: Causes received voice transmissions to not play while the mic key is down.

Always on Top: If selected, the Client window will always remain on top (covering) all other windows.

MIC Sounds: Allows the turning on or off of microphone key sounds.

Sound Quality: Shows the currently selected sound quality settings on the Server.

Output Speakers: Allows the selection of which computer sound card to output sounds on.

Input Mic: Allows the selection of which computer microphone will be monitored for voice capture.

Squelch Level: Allows the selection of microphone sensitivity for squelch mode (voice activation).

Squelch Delay: Allows the selection of what length of silence will disable the microphone under squelch mode.

Packet Time: Allows the selection of the interval time between when UDP voice packers will be sent out.

Buffer Time: Allows the selection of how long the Client will buffer incoming voice data before playing it through the speakers.

Encryption: Shows if the system is currently requiring an encryption key to be input. A “yes” or “no” value is shown. If “yes”, enter the encryption key provided in the encryption key text box and press the “Apply Key” button.

Encryption Key: Enter the encryption key to be used here. This must be a non-empty value to be valid.

Simulate Traffic: Allows simple setup and stress testing of the system by periodically randomly transmitting data over the network.

6.0 Technical Information

All voice data is transmitted via UDP packets with a packet interval selected by the user (60 milliseconds is the default). Transmissions occur on a small subset of the ports numbering between 50,000 to 50,200. The default voice quality used is CELP (Code Excited Linear Prediction) compressed at 32 kHz, 16 Bit, Mono, Variable Bit Rate, but can be set to much higher compression values (with correspondingly lower data rates) as needed. The default voice compression corresponds to an average data rate of approximately 5 kilobytes per second (5 kB/sec), which should be well within the capabilities of both dialup and high-speed Internet access for communicating with large numbers of connected participants. If UDP packets are dropped, due to slow connection speeds or Internet congestion, voice quality may suffer slightly. Under normal conditions this should be unnoticeable.

7.0 Full Duplex Communication

You can set up the system to have an ongoing duplex conversation with one other user without having to constantly press and release the talk button. To do this, set the lock button to hold the talk button down. Transmit on a different channel than the other participant is transmitting on, and monitor the channel the other participant is transmitting on. Likewise, the other participant should monitor the channel you are transmitting on. This setup will cause simultaneously transmitted packets not to get mixed up so even if both parties are talking at the same time, they will be able to hear the other party clearly.