An HF Roundtable with a Skype Based VoIP Bridge

Combine HF and computer based communication to get the message through.

Carl Ferguson, W4UOA, John Krupsky, WA5MLF, and John Drum, W4BXI

A previous *QST* article by Ray Jacob, W2RJJ, gave us an excellent overview of how EchoLink (**www.echolink.org**) can be used to extend an HF net using a voice over Internet protocol (VoIP) bridge.[1] Rarely does HF propagation provide net or roundtable participants with uniform signal strength and readability. The idea that the Internet could allow participants living too close or too far to join a net or roundtable is most attractive and could be a real plus for flexible EmComm operations.

Using W2RJJ's work as a starting point, several amateurs from the Bell Ringers Net established an EchoLink server using W4UOA's base station computer, a microHam microKEYER interface and a transceiver.[2] Early testing was successful. Audio quality was acceptable and EchoLink's support of COM port keying made transmit control easy. Managing the flow of the received HF and Internet audio did prove problematic, however. The varying HF noise floor required the control operator to continuously adjust EchoLink's squelch to control the EchoLink server's transmit function.

EchoLink also proved more than a little challenging for some users. To use EchoLink, one must create port exceptions in a computer's firewall, and not infrequently, port routing on home networks with multiple PCs. Each of these steps is required to allow a participant's EchoLink client to communicate with the central EchoLink server and other EchoLink users. The same steps are required to route EchoLink traffic to a specific computer if the participant is part of a home network.

It was not long before the discussion turned to alternative options for VoIP transmission. Recalling an earlier work on remote control over the Internet, I quickly remembered Skype (**www.skype.com**).[3,4] In short, initial tests of a Skype based HF VoIP Internet bridge were overwhelmingly positive.

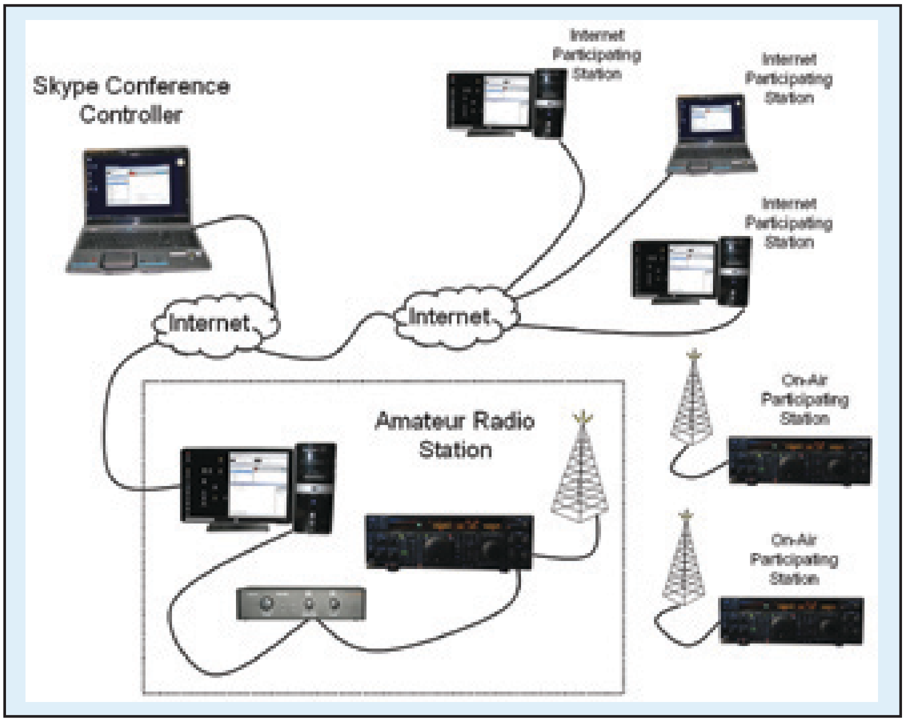

Figure 1 — Pictorial representation of an HF roundtable using a VoIP bridge.

Figure 2 — The W4UOA operating position with the *Skype* software running. The laptop is handling the conference controller function while the base station computer is providing the interface to the transceiver

A Skype Based HF VoIP Bridge

Many *QST* readers are already familiar

[1]Notes appear on page 42.

Figure 3 — The *Skype* login screen — here's where you start.

with Skype, a commercial Internet based long distance voice service. Many users characterize Skype as simple to install and as having robust audio quality. In addition to Skype's ease of use for one-on-one Internet audio and text chat, a Skype client can host up to 25 conference participants. Figure 1 shows a pictorial representation of an HF roundtable with a VoIP bridge. The W4UOA operating position with the *Skype* bridge in place are shown Figure 2.

While such a facility can be assembled and managed in several different ways, the following paragraphs give a step-by-step setup of one successful configuration.

System Components

The current configuration uses an FT-1000 MP Mark V, a microKEYER interface from microHAM (**www.microham.com**), a base station computer running *Windows XP Pro* with all updates and patches, and *Skype 3.8.0.154*. The Skype conference controller is a laptop computer running *Vista Ultimate* with all updates and patches, *Skype 3.8.0.154* and a Logitech USB headset with boom mic. The base station and the laptop share a local telco digital subscriber line (DSL) Internet connection.

We suggest Internet participants install the latest release of *Skype* software and use a headset and boom mic of their choice. We also encourage first time Skype users to use Skype's test call feature to check and set their audio levels. The *Skype* login screen and Skype test call contact are shown in Figures 3 and 4 respectively. Once they install *Skype* and log in for the first time, they should communicate their Skype user name to the control operator. These user names will be used to conference the Internet participants into the Internet bridge.

Sequence of Operation

The control operator tunes the transceiver to the desired operating frequency and advises on-air participants that Internet participants will join the roundtable shortly. At this point, the control operator turns the AF gain on the HF transceiver to zero.

Start Skype *on the base station computer and login to the Skype network.* We suggest setting Skype on the base station computer to answer calls only from individuals who appear in the contacts list and to auto-answer all those calls. In the current application only the conference controller will be allowed to connect to the base station computer.

Start Skype *on the conference controller and login to the Skype network.* A second computer to act as a conference controller is necessary because the base station computer can only run one instance of *Skype*. *Skype* on the base station computer links the radio to the soundcard to provide the A to D conversion on the outbound receiver audio and the D to A conversion of the inbound Internet audio.

Use Skype to place a call to the base station computer on the conference controller. Thus the base station becomes the first conference participant. When the base station computer answers, transceiver audio will be heard on the conference controller's headset.

The control operator enables VOX on the transceiver.

Using the conference controller's headset and mic, the control operator should now confirm the transmit and receive audio link to the base station computer. This can be done by contacting several of the on-air participants from the conference controller. Use these contacts to adjust VOX sensitivity and delay as well as transmit and receive audio levels as necessary. At this point, everything

Figure 4 — The *Skype* test call setup screen. This allows the setting of audio levels off the air.

is in place to bring in the Internet participants. The process is very straightforward but a little practice helps.

Working within the base station connection on the conference controller, ADD+ *each Internet participant.* To do this, each participant must already be in the conference controller's Skype contact list. Remind each participant to follow standard communications exchange protocol to ensure proper identification at all times as well as orderly net discipline. Also remind them that they are on an open VOX line and to use Skype's MUTE button as they would a PTT.

Observations on VoIP Based Operations

Our experience suggests that:

- You should practice using Skype in a conference mode — it's just like learning to use a new rig. You will get increasingly comfortable *managing* the HF Internet bridge as you become increasingly familiar with the Skype's menu structure and command buttons.
- New Skype users may find this link useful as they set up their individual systems: **www.skype.com/help/guides/soundsetup.html.**
- It is best to bring Internet conference participants in by calling them from the conference controller as shown in Figure 5 . Encourage interested participants *not* to call you. If they were to call you, you would have to first connect to them one-on-one and then bring them into the conference. It's much easier for the conference controller to bring them directly into the conference.
- You will have to periodically remind Internet conference participants to mute their mics while not talking.

In addition to *Skype*, our roundtable uses two additional applications: *addonChat* and Adobe *ConnectNow* as shown in Figures 6 and 7. Both are available at no cost and add a great deal to the vitality of the roundtable.

A program our group calls the Chat Room, *addonChat*, provides a reliable real time off-air communications path (see Figure 6).[5] It is excellent for fills as well as additional information to augment on-air discussions. In addition to providing a excellent way to exchange Internet links (URLs) and additional details, participants also use it to enrich the already friendly on-air conversation.

ConnectNow (see Figure 7) provides an easy way to share real time video between participants.[6] Up to three users can log in concurrently and share a live video feed from their respective locations. Frequent uses include showing new equipment, equipment layout and other additions to the shack.

Good luck! We have found the *Skype* based HF VoIP bridge, the Chat Room (*addonChat*) and Adobe *ConnectNow* to be great additions to our early morning roundtables.

Notes

[1]R. Jacob, W2RJJ, "Integrating EchoLink into a Single Sideband Net," *QST,* Aug 2008, pp 52-53.

[2]The Bell Ringers (**tparca.org/bellringers/**) are a group of retired and active Bell System employees and friends that gathers weekday mornings for an informal net and on Saturdays for a formal net. On weekdays they can be found on or about 3968 between 0730 and 0900 CT and on Saturdays at 0900 CT on 7230.

[3]C. Ferguson, W4UOA, "Remote Control Over the Internet (RCoIP)," *QST*, Feb 2006, pp 62-63.

[4]See **www.skype.com/intl/en/allfeatures/conferencecall/**.

[5]See **www.addonchat.com**.

[6]See **www.acrobat.com/#/connectnow/ConnectNowBegin**.

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John Krupsky, WA5MLF, was first licensed

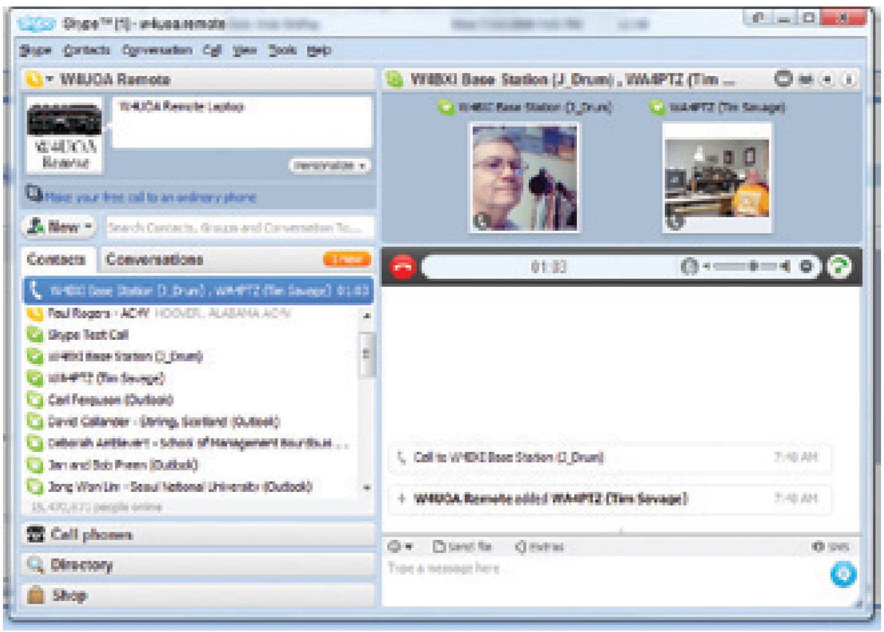

Figure 5 — Screen shot of the conference controller. Skype is used to manage the audio between Internet participants and the base station.

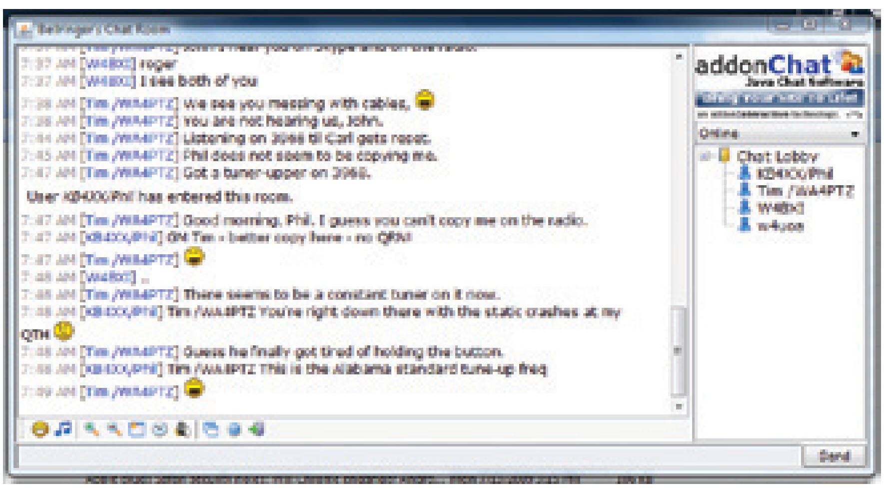

Figure 6 — The *addonChat* chat room provides what we think of as backroom communications between the on air and Internet participants. The chat room is great for fills and comments on the on-air discussions.

in 1965 while in high school. He holds BS and MS degrees in electrical engineering, and served in the US Navy Submarine Service prior to a career in telecommunications. A licensed professional engineer, he retired from BellSouth Telecommunications in 2006, and remains an active member of IEEE. John's ham radio interests include vintage Heathkit equipment, FM repeater maintenance, electronics tinkering, ham-related software applications and soft-ware defined radio. You can reach John at **wa5mlf@arrl.net***.*

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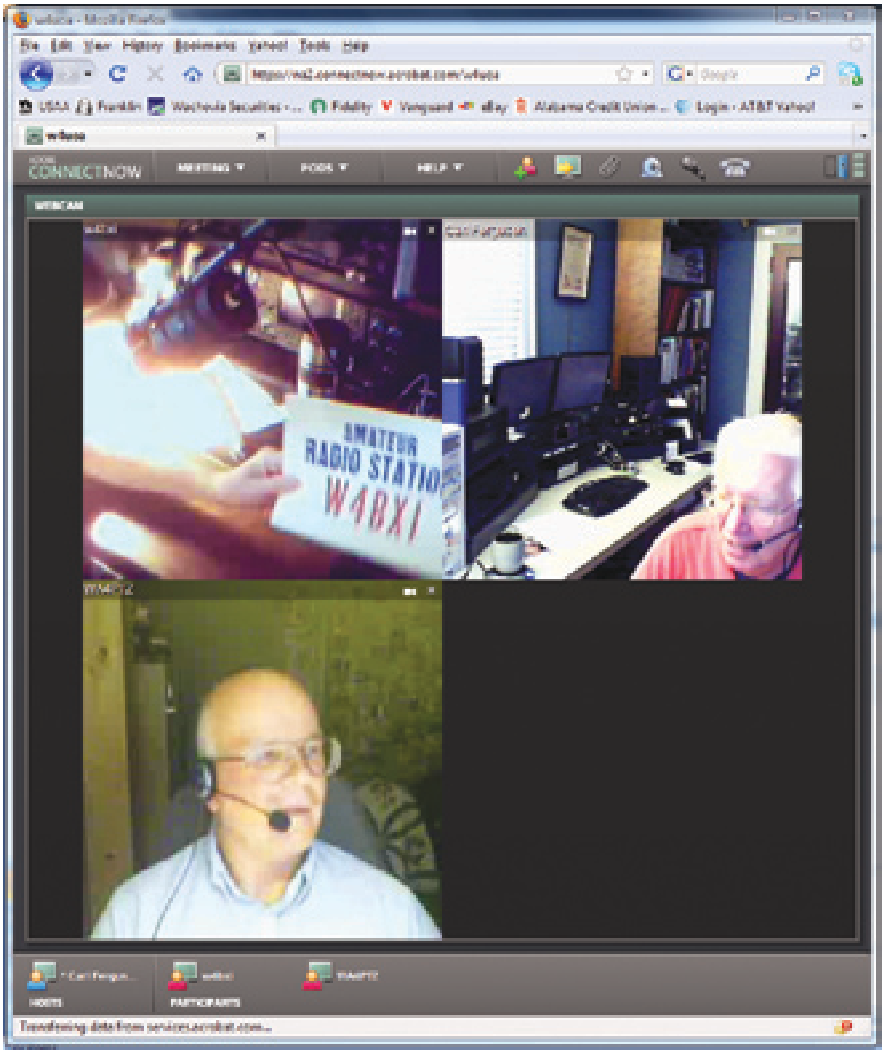

Figure 7 — Adobe ConnectNow allows up to three simultaneous video feeds and numerous other resources. As with *addonChat*, Adobe *ConnectNow* is a available without charge.

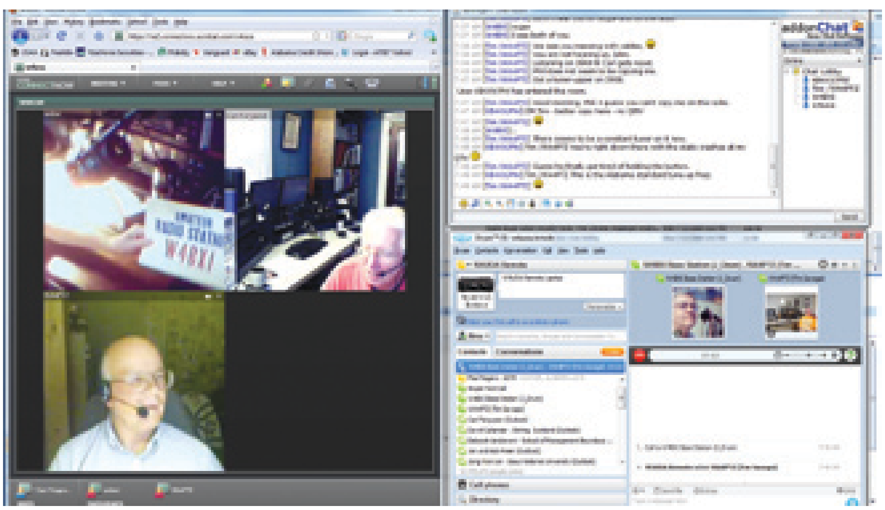

Figure 8 — Real time view of conference controller desktop.

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