Feature Checklist

Check mark means it has been implemented already.

PROOF OF CONCEPT:

P2P HQ mono true-duplex audio direct from one computer to another via Opus codec.

No noise reduction or echo cancellation, by choice.

Solid networking built from the ground up.

Extensible for other protocols.

Drop-down menus to pick audio input, audio output, buffer settings and codec settings.

Text chat for troubleshooting.

Saves settings (including remote IP) on exit.

Buffer meter

BASIC REQUIREMENTS:

Low-latency

No creep in latency.

Roll separate server module into program.

Multi-protocol support (TCP, UDP, Telnet)

Master volume VU meter

Additional audio format / codec: PCM, G.722.

Minimal clicking and drop-outs.

Doesn’t crash.

Works almost flawlessly for two hours of solid two-way talk.

Ability to add third person to conversation.

More variations in Opus settings.

Default to best buffer settings on all three sliders.

Input VU meter for each host ✓

Volume sliders ✓

Default to Opus settings: Sample rate 24 kHz, Bit rate: 32 Kb/s, *Super Wide Band (12 kHz)* but variable. ✓

Remove codec settings that don't test well or aren't needed. ✓

Decide proper license. ✓

Make buffer reset button a little bigger, and easier to see. ✓

Label buffer reset "Buff Dump" (radio broadcasters will confuse simply "Dump" with a cuss-dump button). ✓

Headphone icon on Audio Out section ✓

Hide little-used options, make accessible with Advanced Options button ✓

About Tab with: Readme, manual, credits, BSD3 License, donate link, BTC address and web links. ✓

Basic icons and minimalist branding. ✓

ADDITIONAL GOALS FOR 1.0 RELEASE TO PUBLIC:

Extensive testing completed, bugs worked out. ✓

Works flawlessly for two hours of solid two-way talk. ✓

Well-written, easy to understand manual. ✓

Create installer, package for public use. ✓

Code added on GitHub ✓

GOAL LIST FOR FULL SECOND ROUND OF FUNDING:

Automatic port forwarding. ✓

Make volume slider on Audio Out work more smoothly. ✓

Switchable auto-answer function (with password protection). ✓

Port randomizing option (within a range). ✓

General overall improvements in all functionalities. ✓

Ability to mute and block persons in conversation. ✓

Ability to block IPs. ✓

Add call clock. ✓

Toggable gentle EQ boost at 12hz

IP address display, and ability to copy IP address to the clipboard.

See what else we can move to advanced options.

Add a “Phone book” to save different IP connections and add name or label.

Add ability to save different configurations name them so you call them up easily.

Option to auto-adjust settings for best sound and lowest latency depending on network.

Encryption.

IPv6 support.

Cough button.

Audio alerts on incoming “call” (DJ’s “FeenPhone, FeenPhone” sample.)

Audio alerts switchable to off.

Auto-reconnect if the connection drops.

Input / Output level sliders

Ability to use stereo.

WASAPI support for Audio In.

Ability to roll over to a second pre-set IP if first connection is lost.

Recording (WAV, 16-bit 44.1k mono and stereo).

Mix Minus (this would be a holy grail, it would make this a killer app by replacing audio hardware).

Automation interface with relay closures

PAN left and right: for recording two-host shows for later mixing.

WebRTC support (for callers calling in to a radio show or podcast).

Pro interface and website.

No clicking or drop outs.

Windows Phone version.

WISH LIST OF FEATURES ADDED BY OTHERS, VIA OPEN-SOURCING:

Linux version.

Mac version.

Android version.

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