FeenPhone

Full manual for Installing and Configuring FeenPhone:

FeenPhone has been tested on Windows 7 and Windows 8. It works well on both. It's been used on many different computers effectively. It's even run well on a Windows Surface Pro 2 tablet. FeenPhone will run best on fast computers with 4 gigs or more of memory.



If you want to hire us for tech help with FeenPhone or networking or audio issues, go HERE.

Headphones

You MUST use headphones with FeenPhone when doing two-way audio. And they should be closed-ear, not open-ear headphones (you might already own some). After you test FeenPhone with whatever headphones you have now, check out "Michael W. Dean's 20-Minute Audio School", for more info, including a list of great, inexpensive closed-ear headphones. And you really should read that page to get perfect audio from FeenPhone. Even if you have broadcast experience, but especially if you don't.

Also make sure that your audio is only coming through your headphones, and not also coming through your computer speakers or other speakers you have hooked up to your computer (including speakers in your monitor).

Internet Connection

FeenPhone is pro software for high-quality audio. It is more dependent on a solid, dedicated Internet connection than mediocre-sounding consumer-type VoIP programs. Professional media organizations use expensive hardware & software over dedicated single-use Internet connections for connecting live spoken audio for live radio across the miles. You can get close to that quality for free with FeenPhone, but you can't be downloading huge files at the same time on your home Internet connection. For best broadcast-quality results, don't be doing anything else on your computer or network while using FeenPhone.

I'm usually on a mobile phone over speaker with the other person the first time I help them set up FeenPhone. But there is a text chat box at the bottom right of FeenPhone. This is useful for communicating back and forth while setting up, configuring or troubleshooting. It can also be used for sending show notes and links back and forth during a live show. To use, simply type your message in the box at the bottom. Hit Enter on your keyboard to send the text message.

Installing and Using FeenPhone

-Completely exit Skype, TeamSpeak, Blink, Mumble, Ventrilo, Google Hangouts and any other Voice-over-Internet programs you have running.

Skype still runs in the background when you close the program. By default, you have to take additional steps to close it. (We will NEVER add this "feature" with FeenPhone.):

-Close the Skype program.

⁻Go into the system tray and click "Show hidden icons":

-Right-click on the Skype icon (circled in red below):

-Click "Quit"

Depending on your computer's settings, you may or may not get a Windows Firewall warning or antivirus program warning when downloading, installing or running FeenPhone. There is no virus in FeenPhone, it's simply too new to be recognized. If you do get warnings, chose "Allow" or "Allow Always" to bypass the warning. (In Windows Defender, you have to hit "More Info" on the notification window to get to the place you can chose to run anyway.)

 – DOWNLOAD FeenPhone beta version. (If you had another version of FeenPhone installed, uninstall it first.)

-Click on the downloaded installer, and let it install.











Click "Next." It may take up to two minutes before it looks like it's doing anything, but then you'll get a series of "Next" windows to click through. Click "next" through the dialog boxes until it's done.

-If you get a Windows prompt saying that your Windows .NET Framework is not up to date, allow it to update.

By default, the FeenPhone installer will add a desktop icon. If you want, you can right-click on the desktop icon or the program icon from the Windows "All Programs" listing, inside the FeenPhone folder, and pin a shortcut to the task bar. You'll probably want to, since FeenPhone is so awesome, that if you do a lot of podcasting or radio or voiceover, you'll be *living* in FeenPhone.



-Open FeenPhone:



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FeenPhone just after being opened (your Input and	Output devices will	be different)		

FeenPhone starts with the AUDIO OUT button (microphone icon) muted (red X showing). Once an audio input is connected and chosen, and you connect to another user, the X should go away.

Name

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The Nickname field will say CHANGE ME by default. Enter your name (or a user name) in place of that. When you start typing in that box, the CHANGE ME will go away and be replaced by your name. Two or more users on the same call cannot use the same user name.

NETWORK Settings:

When two or more people use FeenPhone, *one of them has to act as the server, and will have to configure their router to forward a port* for TCP (not UDP) in the 5000-6000 range. Directions for forwarding a port are here. There is more router model-specific port forwarding info in this Bittorrent-support page.

-The computer running as the server ("the Server") does not need to enter anything in the Server field. They do need to enter their open port number in the Port field (to the right of the circled red button in the image below). The other computer(s) connecting ("the Client" or "the Clients") will need to enter the IP number address and Port number of the Server, in the Client tab. The Server can find their own IP address to give the Client(s) by clicking this website or by Googling "My IP". (The Server address can also be an alphabetical web URL rather than an IP, if you know how to set that up on your end.)



It doesn't matter which person's computer runs as the Server, but the computer running as the Server must push the "Run Server" button on the Server tab near the top right of the program interface (that button is circled in red below) before the Client(s) push the Connect button. The Client(s) should NOT push the Run Server button.

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You can tell when the Server is running because it will say "NETWORK: RUNNING SERVER" near the top right of FeenPhone:

The Server will know the Client is connected when the Server's LOG field says "Login SUCCESS: (Client user name)."

The Client will know they're connected to the Server when the Client's LOG field says "Server Login Accepted" and shows ping response times.

Changing the Port

The default port when you open FeenPhone is 5150. You can use that for initial testing, but you should change to a different port before using FeenPhone for mission-critical live



media. (Since it is public info here, that would likely be the first port people would try if they were attempting to monkey

around with your connection.) Do not give out your forwarded port number to the public.

The Server port in FeenPhone is changeable whenever the Server is not running. If you've been running as Server, stop running the Server on your FeenPhone by clicking Run Server in the Server tab. Change the port number in the Server tab to another port you have forwarded in your router. Then click Run Server again, and the Client should be able to enter your new port number and your IP address and connect.



DSL lines tend to issue a new IP address every time the

modem is re-started, so if you're on DSL, you can usually just re-start your modem to change your IP. Cable Internet modems usually have to be off for a day or two to issue a new IP address (unless you're paying extra for a dedicated IP, then you have that IP and will have to contact your provider to change your IP). DSL has much lower bandwidth than cable Internet, but it's more than enough for FeenPhone. And DSL is much more solid than cable. Cable is shared with all your neighbors, DSL is a dedicated line. I have both, and always use FeenPhone over DSL because it's more reliable. I keep cable for general Internet use and as a backup. Another tip: FeenPhone will run much better if you reboot your computer at least once a day.

AUDIO IN settings:

FeenPhone starts with the volume sliders halfway up or all the way down, depending on your system. You may have to change your volume sliders if it's too loud or too quiet, depending on how loud you are to the other person, and them to you.

The input and output volume sliders in FeenPhone control the volume slider in your computer's System Tray (bottom right of computer desktop). When you move the slider in FeenPhone, the slider in the system tray opens. Some sound cards have their own program for controlling their volume. If you can't move a slider, this is probably the case and you'll have to control the volume in and/or out from that device's proprietary volume control program in your computer. If you still can't hear your co-host, or your co-host can't hear you, go to the Troubleshooting page.

The sliders are labeled by user name for each person (to the left of the mic icon and headphone icon). There's also a MASTER volume slider that controls all callers, including you. And each remote Client, plus the MASTER volume, have a VU meter (Volume Unit meter):



You can use the VU meters to keep an eye on everyone's volume. (A VU reading averaging around 1/2 to 2/3 up wher people are speaking is a good place to shoot for.) The image below shows a good volume on the MASTER volume VU meter and the remote Client's VU meter (the one labeled "derrick"). (This is on a three-way call, three-way calls are explained further down this page):



For now, the AUDIO OUT's MASTER volume slider will only work on WASAPI settings in AUDIO OUT. We will expand this in an upcoming version to work on any AUDIO OUT setting.

All FeenPhone settings can be changed on the fly. FeenPhone will save your basic settings upon exit, and they will be there the next time you start the program. In the future we plan to add a "phone book" sort of interface so you can save several named contacts and their IP/port settings so you call them up easily.

FeenPhone's pre-set defaults should work in general, but some things need to be configured if they're not set to your system's needs.

While trying different settings, both (or all) people should be talking, so you'll know when it works. Sometimes one person can hear the other person, but not vice versa. The text chat function in the bottom right of FeenPhone is useful to alert the other person that something is working, at least in one direction, if you can't both hear each other.

-In the AUDIO IN drop-down menu, pick your input device. If you're using the recommended Audio-Technica AT2005USB mic (microphone) (Get it here on Amazon), it should show up in the drop-down list as something like *WAV: Microphone (AT2005US)*:



If you're using a different USB mic, look for the model number of your mic in the drop-down list and pick that. If you're using a mic or mixer that goes in through your sound card (or sometimes even if a mixer connects via USB), you may have to pick your mixer, your sound card or your general USB interface.

The AT2005USB mic's blue light will come on whenever you are connected to your computer via USB. But you still need to turn the mic's switch on (switch pushed toward the talking end of the mic) to get audio! The blue light will come on even if that switch is off. There's more info on this mic on the page "Michael W. Dean's 20-Minute Audio School." And this is kind of a no-brainer, but I've seen it done wrong so I'll mention it: you should talk into the end of that mic (where the flat part of the wire mesh grill is), not into the side of the grill!

-Default for the Input Buffer is 70ms. Should be good. You will often get audio drop-outs or clicking sounds when FeenPhone first starts, then the network will adjust and those glitches will go away in less than a minute. If you continue to get drop-outs or clicking, you can increase the Buffer Target slider a little bit. This will give more-solid sound, but may give a noticeable amount of round-trip latency. If this doesn't solve the issue, have the person you're talking to try turning the Buffer Target slider up. If that doesn't solve it, see the Troubleshooting page.

If both users have great computers and a great connection between you, you may be able to even *lower* the Input Buffer for less latency and still not get audio glitches.

More buffer means more latency, but higher audio quality. All the buffer sliders are a trade-off between having better audio quality and getting a low enough latency to have a natural, conversational feel where the other person doesn't wait too long before hearing you and replying. You may have to experiment to get the best audio with the lowest latency for your system and network. And it may vary with the system and network of the person you're talking to. But again, FeenPhone's defaults are a great place to start.

If you're doing one-way transmission and latency is not as much of an issue, like with a studio-to-tower link, you could turn the buffer higher. If synching radio reception live to an atomic clock is not an issue, you could have both buffer sliders all the way up (all the way to the right).

The correct settings are going to vary a lot based on the mic /sound card / network. We've designed FeenPhone's defaults to work with the Audio-Technica AT2005USB mic over a solid cable Internet or DSL connection, but your mileage may vary. There are many variables that can interact differently with combinations of various computers, sound cards, microphones and networks. Computers are an interesting mix of science and voodoo.

AUDIO OUT settings:

-Set your Output Device.

If you're using the recommended Audio-Technica AT2005USB mic, it should show up in the AUDIO OUT drop-down list as something like WASAPI: Speakers (AT2005US). (See next image up, AUDIO IN and AUDIO OUT in FeenPhone for AT2005USB mic.)

This is because you plug your headphones into the bottom of the mic, so the mic is your output device as well as your input device. There's also a volume control on the bottom of the mic, it only controls your headphone volume, not the volume of your output.) If you're using a different USB mic, that mic will probably show up as Input, and as Output if it has a headphone jack on the mic like theAT2005USB. If you're using a USB mixer or a mic plugged into a sound card, your Output device will likely be your USB mixer or sound card. (It will NOT be the model of your headphones, unless they're a USB headset.)

If you get two options for your AUDIO OUT device, pick the one with "WASAPI" in the listing, it will probably sound a little better.

You can plug your headphones into your computer instead of the bottom of the AT2005USB, but you won't be able to hear yourself, you'll only hear the other person. Some people like to hear themselves, some people don't. If you plug your headphones into an audio output of your computer, you'll probably see your computer's sound card as the proper Audio Out device. But if you have a mic with an on-board headphone out, we recommend plugging the headphones into the mic so you can hear yourself. You'll develop better mic technique more quickly if you can hear yourself.

More info on mic technique is in Michael W. Dean's 20-Minute Audio School

Buff Dump

The Buff Dump button will empty the buffer. It's useful if you start getting drop-outs, clicking or too much latency. Some buffer is good, like averaging around the first fifth of the bar. But if it regularly goes over half-way (or if you hear too much clicking), hit the Buff Dump. I generally hit the Buff Dump between two and five times per hour on an average Internet connection, and almost never on a great connection. Either person can hit the Buff Dump, but should not do it while the other person is talking or there will be a drop-out. Hit the Buff Dump on your end only when you are talking, or when there is silence.

Even with everything optimized, you may hear a tiny amount of quiet clicking from time to time. It's certainly no more prevalent than with other VoIP software. The very occasional clicking issue in FeenPhone is something we plan to deal with in later versions, funding permitting.

Since FeenPhone has no echo cancellation or noise reduction, by design (those things kill great audio), it is possible that you'll get sound leakage from your headphones into your mic. Be sure to use closed-ear headphones (not openear headphones or ear buds). If the other person hears themselves quietly echoing back, turn down your headphones. Also, don't have your hair under your headphones, it will create a gap and let sound echo through into your mic.

Mute Button

You can click on the mic icon in FeenPhone to mute your mic if you need to cough or make a noise, but it will give an abrupt drop in any background noise when you do, and you'll need to click it again to un-mute it. You'll see a red X when you are muted. We will be adding an actual cough button later, that will not make an abrupt drop, and will only mute while held down.

There you go. Happy Feening!

Networks

You should have your computer WIRED, not WIRELESS, you'll get a more-solid connection. If you have roommates or anyone sharing your Internet connection, they should not be streaming media, doing large downloads or torrenting while you're doing a show. And you should not be doing high-intensity functions on the same computer that's running FeenPhone.

FeenPhone will work over a 4G network, but not nearly as well as over a wired Internet connection. FeenPhone will barely work over a 3G network. FeenPhone will not over a VPN or Tor. There will be too much latency for a conversational back-and-forth feel.

Don't run your USB mic or USB mixer through a USB hub. Connect the mic or mixer directly into a USB port on your computer.

When doing a live show, we recommend using a dedicated computer for FeenPhone. Even a cheap used computer running FeenPhone alone will usually perform better than using FeenPhone on a better computer that's used for other tasks during the show. You can get a used Windows 7 laptop or desktop on eBay in a "buy it now" sale for about 100 dollars. FeenPhone is such a killer app that if your only computer is a Linux box or a Mac, instead of complaining that we haven't made a version for you yet, you might consider getting a used Windows machine just to run FeenPhone. A couple of the Freedom Feens co-hosts have done that.

Testing FeenPhone Without a Second Person

You can run two instances of FeenPhone on one computer, one running as a Server, one connecting as the Client. Just click on the shortcut twice to open two instances. You can even run more than two instances, with one as the Server and the others connecting as Clients. For the "Server" field on the Client instance, enter "localhost".



We will also be setting up a dedicated echo server with public IP/Port info that you can log into to check FeenPhone in at least in one direction. The server on our end will be playing a continuous loop of Freedom Feens episodes. We may actually allow you at some point to connect to anyone else who logs on at the same time, a free "FeenPhone Party Line." We'll announce these on the FeenPhone blog when it happens. Subscribe to our RSS Feed for updates on that, new versions, and all the fun FeenPhone news.

Passwords

FeenPhone's default is passwords off, i.e. anyone who has your IP address and port number can join your call when you're acting as Server. But you can enable the requirement of using a password to connect. (Later versions will add ability to block IPs, and will add encryption. There is none for now.)

To enable passwords, you have to be the Server. While not connected to anyone else, click on the Server tab, stop running Server if it's running, and click the tick box labeled *Require Auth*:

You will receive a prompt to set a password:

Type the password you want to use, and click "Set Pass":

When the Client enters your IP address and port, they will be prompted for a password. Without the password, they will not be able to log in.

	ABOUT
	NETWORK: RUNNING SERVER
•	Client Server
	Run Server Ports 5150
	Require incoming clients to authenticate
•	USERS

Sometimes the Client will be prompted to enter the

password again and hit Login again. To disable requiring passwords, uncheck the "Require Auth" tick box on the Server. To change the password, click the "Manage Authentication" button on the Server.

The Client can disconnect and reconnect without re-entering the password, but the password will be cleared when the Client is closed. The Server will save the password when closed.

Three-Way FeenPhone Use

You can connect as many people as you'd like on FeenPhone, but on most systems it won't work well with more than three people. (More people will cause more clicking and audio glitches.) We don't plan to work hard at making FeenPhone work with more than three people. It technically will, but each person adds latency and demands on the Server's processor, so more than three will not be promised. But more than three people not in the same room talking is a ALWAYS a clusterfark anyway. Four people on a show barely works in the same room, and not at all from a distance. Any latency, combined with not being able to see each other for the visual cues of when a person is about to start or finish talking, makes it not work. This is always true, with any VoIP program. It's even true if the people can see each other via video. Look at any four- or five-neckbeard Google Hangouts show on YouTube for examples.

Remote shows require some practice. They require paying attention to and actively listening to the other person while you're talking. This is much more possible with FeenPhone, because it's truly duplex, i.e. both people can talk at the same time without cutting each other off, unlike other consumer VoIPs. FeenPhone's true duplex and lack of noise reduction lets you hear subtle cues in people's talking, like when they take a breath or just barely start a syllable.

Three-way remote shows require even more practice to do well.

With three-way calls, a second set of meters, Buffer Target sliders and Buff Dump buttons is added below the first one:

Feen Phone: Set Server	Password.
Set Server Password.	
Password	
Cancel	Set Pass

 Feen Phone: Set Server Password. Set Server Password. Password Cancel Set Pass
Feen Phone:
Password Cancel Login
Client receives password prompt
Gen Phone:
Password ••••

Client types password and hits "Login."

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You can use these sliders as a mixer to balance one louder person and one quieter person to be the same volume. You can't do that in other VoIP programs! You can also use the MASTER volume slider on FeenPhone to control the volume of all the remote hosts, and you, together.

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ADVANCED SETTINGS

A lot of the Advanced Settings in FeenPhone are hidden. The average user will never have to deal with most of them. If you want to check them out and play around with them, click the little blue bird icon in the top right corner to the left of the About button (click image below to see larger):



Buffer Recovery on/off: You'd probably only ever turn this off for advanced testing if you really know codecs and networking and such.

Codec Picker. You can each use whatever codec you want, but *we've generally had the best results with everyone using the Opus Talk setting – 24khz (32kbs)*. Server and Client do *not* need to be on the same Codec setting. You can see what the other person is using, their codec will show up under "Codec" in your Audio Out section.

Basically the Opus Music setting is in there for use as a radio station signal-to-tower link on very high-quality dedicated Internet connections (for music or for talk). Try it on both Server and Client for voice if you and your remote co-host both have rock-solid Internet connections. It will sound even better than Opus talk. But if your connections are anything but perfect, you'll get clicks and dropouts.

The G.722 and PCM settings are for later expansion to connect with hardware codec broadcast boxes....like a Comrex, Tieline or Telos....So you can even use FeenPhone to connect to a radio network without additional equipment, should you get syndicated. (Though my radio network will be allowing us to connect directly via FeenPhone once we're out of beta. Other networks are likely to follow.) And if you get an inexpensive mixer and hardware compressor, you and your host can sound even better!

Wave / WASAPI picker: For now Wave works best on the Input, and WASAPI works best on the Output. In a later version of FeenPhone, WASAPI will be added for the Input, and will likely work better than WAV.

Protocol Picker: Many people think UDP works better for VoIP. We designed FeenPhone to work better with TCP. TCP is more reliable, due to its error correction, higher priority in most routers, and guaranteed delivery of packets. But you can switch to UDP if you want. (You'll have to open the port you're using on the Server end to allow UDP in order for it to work, and enter the port into the UDP port section in the Server tab while in Advanced Options, see below.)

Other Protocol Port options (Server tab only): allows you to enter ports for UDP and even Telnet if you know what those are and want to experiment. (UDP will work for voice, Telnet will only work for text chat.)

Known Issues in FeenPhone (All will be addressed in later versions.) –Some clicking sounds no matter what

the buffer setting. -Very rarely the chat box stops working. If this happens, a re-start of the program will fix it.

-Occasional crashing (usually the fault of not exiting other VoIP programs first.)

Uninstalling FeenPhone What What What?! Are you crazy?! Why would you do that?

OK, if you *do* have to uninstall for some unimaginable reason, go in Windows to Start / Control Panel / Programs and Features. Find FeenPhone (look for the yellow icon),

FeenPhone (look for the yellow icon), click it and Windows will uninstall it. Or go to Start Menu / All Programs / FeenPhone program group, and click "Uninstall."

Redemption in the Third Act

If you read all of this carefully and you're still having issues with FeenPhone, post a comment below. Be sure to include as much info as possible: version of Windows, computer speed, network type, etc. (Don't just say "IT'S NOT WORKING!", we won't be able to help you!)

If it's a simple issue, we'll try to help you in the comments. If it's complex, we can be hired for remote tech support, audio help or radio consulting here.

And if FeenPhone is working well and you love it, please tell two friends, and consider supporting open-source nocost software by making a donation, here, or by clicking on the About tab in FeenPhone.

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	Advanced	CHAT Settings in Server	Tab	

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	NETWORK		BOOT	
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Check out the About tab anyway. it's pretty nifty, and lists all the people who helped out with the donations, and the folks who helped with alpha and beta testing.

Woims!

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-THE BEASTLICK INTERNET POLICY COMMISSION outreach team