

MicroSIP Configuration Guide

Overview

MicroSIP is a soft phone that is compatible with the IPFINITY CloudVoice[™] service. It functions similar to any other IPFINITY full featured phone such as the Grandstream GXP2135/GXP2170. It includes features such as call recording, touch tones (DTMF) and voicemail.

The purpose of this document is to guide users into configuring their IPFINITY account with MicroSIP.

Preparation

MicroSIP requires the following hardware.

- Windows PC with compatible versions in the next section
- PC with microphone or headset with microphone

MicroSIP is compatible with the following Microsoft Windows versions.

- Windows XP
- Windows Vista
- Windows 7
- Windows 8
- Windows 8.1
- Windows 10

Installation

Download the latest version of MicroSIP from the following URL.

https://www.microsip.org/downloads



For purposes of this documentation, the full version with the installer will be used. The file to download should be named in the format "MicroSIP-X.X.X.exe", where X.X.X is the version number.

If you are unable to install the software, you may need to ask your system administrator to install the software for you.

Once downloaded, run the installer and follow the steps as needed to complete the process.

After the installation is complete, run MicroSIP from the Start Menu and proceed to the next section.

Adding Your Account

This section will guide you into adding your account to MicroSIP. Please follow the steps in sequence.

1. Bring the MicroSIP window to focus and familiarize yourself with the interface.

hone Logs	Contacts	•
		· · ·
1	2	3
4	5	6
7	8	9
*	0	#
R	+	С
۲	Call	Ģ
•		
<u>.</u>		
	DND AA	REC Q.O
MicroSIP		



2. Open the main menu by clicking the button encircled in red as the example screenshot below.

G MicroSIP	-	
Phone Logs	Contacts	
1		•
1	2	3
4	5	6
7	8	9
*	0	#
R	+	С
<u>.</u>	Call	ø
±		- 1
	DND AA	REC QO
MicroSIP		

Figure 2. Main menu button encircled in red

3. Click on "Add Account..." to open the account configuration window.

Logs	Contacts			Add Account Settings	Ctrl+P
1	2	3		Shortcuts	Ctrl+S
4	5	6		Always on Top View Log File	
7	8	9		Visit Website Help	Ctrl+W
*	0	#		Exit	Ctrl+Q
R	+	С			
	Call	F			
4 ·		-	ī		
		REC QO	i I		

Figure 3. MicroSIP main menu



- 4. Enter the information in the following fields. Leave the field blank or as is if not specified.
 - Account Name: ipfinity
 - SIP Server: (refer to your account credentials)
 - Username: (refer to your account credentials)
 - Domain: (refer to your account credentials)
 - Login: (leave blank)
 - Password: (refer to your account credentials)
 - Display Name: (leave blank / set to your preference; may be overriden)
 - Voicemail Number: 00
 - Media Encryption: "Disabled" selected
 - Transport: "TLS" selected

Click "Save" when done.



Account		×
Account Name	ipfinity	
SIP Server	server.ipfinity.com	2
SIP Proxy		2
Username*	XXXX	?
Domain*	server.ipfinity.com	2
Login		2
Password	•••••	2
	display password	_
Display Name		2
Voicemail Number	00	2
Media Encryption	Disabled •	2
Transport	TLS •	2
Public Address	Auto	2
	Publish Presence	2
	Allow IP Rewrite	2
	Disable Session Timers	2
	Save Cancel	

Figure 4. Account window example

- 5. Click on the main menu button again and select Settings. Alternatively, you may press Ctrl+P from the MicroSIP window. The Settings window will pop up.
- 6. Focus on the two item lists labelled "Available Codecs" and "Enabled Codecs". Move the following three items to the "Enabled Codecs" list with the following order.
 - 1. G.722 16 kHz
 - 2. G.729 8 kHz
 - 3. Opus 24 kHz

Click an item on one of the lists to select and highlight. Use the buttons with left and right arrows to move the selected item between the two lists. Use the buttons with up and down arrows to move the item's position in the list. The list in "Available Codecs" does not have to be ordered.



Click "Save" when done.

Settings				
	2 V Sing	je Call Mode		
Ringin	g Sound			x 2
Rin	Device Defaul	t	•	
1	Speaker Defau	t	•	
Mo	ophone Defau	t	•	
	Microphone	Amplification		z
	Software L	evel Adjustment		2
Availat	le Codecs	Enabled	Codecs	
G.711 G.713 G.723 GSM 8 AMR 8 AMR-1 iLBC 8	A-law u-law 8 kHz kHz kHz kHz WB 16 kHz kHz	G.722 1 G.729 8 Opus 24	6 kHz kHz 1 kHz	2
2 🔄 VAD	2 🔽 EC	2 Force Code	c for Incoming	
Call Reco	rding C:\Us	ers\temp\Desktop\P	Recordings	x
	Camera Defau	t		P
	Coder Date		-	

Figure 5. Enabled Codecs list example



7. Verify that MicroSIP is properly configured with the status bar located at the bottom of the MicroSIP main window. A text saying "Online" and a green telephone icon should be displayed.

🕒 MicroSIP - i	ipfinity 📃	
Phone Logs	Contacts	•
		•
1	2	3
4	5	6
7	8	9
*	0	#
R	+	С
۲	Call	Ģ
4		
L		I
	DND	REC 0.0

Figure 6. Configured and ready MicroSIP example



Basic Usage

Attended Transfer

To perform an attended transfer or warm transfer, single call mode needs to be disabled in MicroSIP settings. To access MicroSIP settings, press CTRL+P.

The following example shows how to make an attended transfer of an ongoing call from "JP Loh" to the number (416) 900-1416.

1. Dial the destination number or call from the contacts. Once the second call is initiated, the party being transferred will automatically be placed on hold and hear hold music.



2. Wait for the receiving party to answer the call. Notice that the first call has been placed on hold with the icon changed to the pause icon.





3. When ready to transfer, go to the call to be transferred then click on the Transfer button to open the transfer menu.

				Mana and a state of the second of the second se
😴 JP Loh 📃 🗖 🔤		🕒 MicroSIP -	ipfinity 🗌	- • ×
II JP Loh 🖾 😏 4169001416 📧 Last Call Close All		Phone Logs	Contacts	•
Cell Transfer				-
[10:13:26 AM] Connecting	L	1	2 ABC	3 DEF
[10:13:26 AM] Connected (fc4.ipfinity.com, PCMU@8kh: 64kbit/s)	L			
[10:14:31 AM] Call on Local Hold		4 GHI	5 JKL	6 MNO
		7 PORS	8 TUV	9 wxyz
		*	0	#
		R	+	С
		۲	Call	Ģ
-				
				T.
		DND	C AA CON	F REC Q.O
		S Connected	I (2)	2023

4. Hover your mouse over Attended Transfer and select the call of the receiving party.

		_			Man State
🕒 JP Loh			🕃 MicroSIP -	ipfinity	
🔋 JP Loh 🖾 🕓 4169001416 🖂	Last Call Close All	Г	Phone Logs	Contacts	•
	Conference End Call				
[10:13:24 AM] Ringing (fc4.ipfinity.com)	Attended Transfer > 41690	0014	16		
[10:13:26 AM] Connecting			1	2 ABC	3 DEF
[10:13:26 AM] Connected (fc4.ipfinity.com, PCMU@8kHz 64)	(bit/s)		1.00	E	C NUNC
[10:14:31 AM] Call on Local Hold			4 GH	5 JKL	OMNO
			7 PORS	8 TUV	9 wxyz
			*	0	#
			R	+	С
				Call	Ţ
	-		•		
			1		T.
			DND	C AA CON	F REC 0.0
1		3	Connected	(2)	2023

5. Once the call has been transferred, both calls will end. Each call's tab will remain.





Do Not Disturb (DND) Mode

Do Not Disturb Mode or DND Mode will reject all incoming calls when enabled. Rejected calls while DND mode is enabled will appear as missed calls in your call history.

To enable or disable Do Not Disturb mode, go to the MicroSIP window and click on the DND button in the bottom portion of the window.

When enabled, the status indicator in the bottom will indicate "Do Not Disturb" with the phone icon accompanied by a small "No Entry" sign.

How to Exit MicroSIP

In order to avoid missing calls, closing the MicroSIP window does not exit the application. To exit MicroSIP, right-click on the MicroSIP icon in the system tray then click on Exit. Alternatively, you may press Ctrl+Q on the keyboard when in the main MicroSIP window.



When MicroSIP is not running, you will not receive calls. The system will immediately send callers to your voicemail mailbox if there is no other device to ring.

Troubleshooting

Below are basic steps to troubleshoot MicroSIP with your IPFINITY service.

- 1. Restart MicroSIP and/or Windows. Power cycle your modem, router and any related network equipment for Internet access. Check that any headset cable is securely connected to the appropriate port/s.
- 2. If you are hearing one way audio (cannot hear the other party or the other party cannot be heard), ensure that the speaker and microphone button is not muted (both buttons are below the call button, the button is muted if the icon has a diagonal strikethrough. If the issue persists check your Internet connection, your PC's firewall settings or consult with the system administrator.



- 3. If MicroSIP is not online or calls do not connect, ensure that your computer has access to the Internet and check the firewall settings or consult with the system administrator.
- 4. If audio is getting interrupted with silence, "choppy", or, speech sounds robotic, check your Internet connection and run a speed test. Restart your modem, router and any other related network equipment.
- 5. If there is cracking noise, distorted speech or other possible audio issues, dial 334 from MicroSIP to use the call testing service.
- 6. If MicroSIP is not ringing when the extension is called, or, calls are sent to voicemail without ringing, ensure that MicroSIP is running. Restart MicroSIP or the PC if issue persists. Check that DND is not activated ("Do Not Disturb" in the status bar. Otherwise, check your Internet connection or consult with the system administrator.

For more network troubleshooting, you may consult our Knowledge Base article entitled Network Checklist in the following link.

https://ipfinity.zendesk.com/hc/en-us/articles/227388068-Network-Checklist

To know more about MicroSIP, all its features and help on using its other functions, please visit the following website.

https://www.microsip.org/

For further assistance, consult with the system administrator or IPFINITY by sending an email to support@ipfinity.com or calling (855) 473-4648 option 2.