

## **SIP Instructions**

### Enter a *SIP Ring-To address* on the TollFreeForwarding.com website

1. From the Home page of your Online Control Center, choose Manage Phone Numbers > Phone Numbers



2. Choose "New" under Call Actions

Call Action 🥐	
1234 - David	▼ C Edit C New Test Call Action

3. Next, select "Call Forwarding"



4. Then "Create New Ring-To"

Call Forwarding Destination ?	
12345551212 David Home	
📞 Edit Ring-To	

5. Under "Send Calls To" select "SIP/VoIP" from the dropdown menu



Ring-To	ĺ	🧹 Save	X Cancel
Description David's SIP Forwarding			
Send Calls To			
Phone Number A country code is required Don't use international dialing prefix such as 001, 011	SIP/VoIP SIP/VoIP USA/Canada (+1) Abkhazia (+995) Afobapistap (+92)	<b>▼</b>	-

6. Enter the SIP address that will be receiving your calls

end Calls To	
hone Number A country code is required Don't use international dialing prefix such as 001, 011 Don't use a national dialing prefix	SIP/VoIP   18885551212@ sip. domain.com   SIP/VoIP   Important: If this Ring-To number is a fax number, please disable Fax2MyEmail <sup>74</sup>
lease do not forward this ring-to t prough our network and you will b	o your TollFreeForwarding.com number. This can cause calls to loop e responsible for any charges that occur from this.

## **Examples of SIP Ring-To address formats:**

user@sip.server.com	user part comes before @ server name comes after
8005551212@99.99.99.99	IP of your SIP Server
bob123@99.99.99.99	User can be any string of characters
8005551212@sip.mydomain.com	DNS of your SIP Server
8005551212@sip.mydomain.co.uk	DNS from any country is acceptable
8005551212@A-RECORD.mydomain.com	Most customers use common DNS A Records
8005551212@DNS-SRV.mydomain.com	DNS-SRV available for advanced customers
DIALED@sip.customer.com	Replaces "user" with the number dialed



#### **Ring-To notes:**

- The IP address may have fewer numbers between the periods, but it always has four sets of numbers separated by three periods: @1.2.3.4.
- Unless you are very familiar with DNS, most of our customers find it easier to use the IP address of their SIP server instead of DNS.
- Make certain that the user (all characters before the @ symbol) does not include spaces:

Incorrect:	user name @domain.com	(includes spaces)
Correct:	username@domain.com	(no spaces)

#### Add Multiple Ring-To Numbers (Optional)

Note: Multiple ring-to destinations may be set up to include a mix of SIP addresses and phone numbers.

- Within the call forwarding action, click "Add New Destination".
- Choose the Ring-To number from the drop down list, or click "Create New Ring-To" and enter the new number to forward to.

escription SIP User		Siew Tutorial
ing Delay ?	Call Forwarding Destinatio	on ?
o delay, immediately ring	18885551212@sip.domain.	com David's SIP - ↓ Create New Ring-To
	+ Add New Destination	
■ Enable Voicemail2MyEmail™		
Show Advanced Settings		



Call Failover Setup (Optional) - First ring-to SIP and if your SIP fails, ring-to a phone number

• Set a ring delay for the failover route. If your SIP server fails, it normally does so within less than a second. We suggest setting a 6 second delay.

Call Forwarding	ange Type of Call Action 🖌 Save 🗙	Canc
Description     SIP User       Extension @     (optional)	Providence Street Stree	torial
Ring Delay ?	Call Forwarding Destination ?	
No delay, immediately ring Scontinue Ringing	18885551212@sip.domain.com David's SIP     Lit Ring-To     Lit Ring-To	
(If no answer after 6 seconds, then ring \$)	12345551212 David Home	
	+ Add New Destination	
Enable Voicemail2MyEmail M 2		
- Show Advanced Settings		

## Configure your SIP server to accept calls to the User Part entered in "Ring-To"

# Note: Make certain your server is configured and ready to accept incoming calls to the SIP User Part. This is the most common misconfiguration.

We are happy to recommend SIP settings, but we are unable to configure your SIP servers on your behalf. For assistance with SIP configuration, consult your IT team or the provider of the SIP server.

- Check that your SIP server settings match the Ring-To entered in Step 1
- Create settings for your chosen User Part in userpart@sip.server.com
- This User Part will control how your SIP server handles incoming calls
- You may direct incoming calls to various actions by using multiple Ring-To addresses with matching User Parts



## **Configure your SIP server to receive incoming calls from our IP addresses**

SIP Signaling IP Addresses			
Los Angeles Primary Datacenter	Los Angeles Secondary Datacenter		
52.144.10.100	52.144.11.100		

RTP Media IP Addresses				
Note: Most systems do not need to be configured for these specific RTP Media IP address				
	IP	Netmask	Subnet Mask	Range
Worldwide	52.144.0.0	/20=	255.255.240.0	52.144.0.0 to 52.144.15.255

## **Configure your SIP server to accept at least one of the following CODECs**

Primary CODEC used:

G711uLaw (also called PCMU)

Note: The following TFF Ring-To settings are  $\ensuremath{\mathsf{ONLY}}$  supported by  $\ensuremath{\mathsf{G711uLaw}}$ :

- Dial Foreign Extension (DTMF tone allows the SIP server to connect to a specific extension)
- Auto Answer Prevention (requires a physical key press to accept incoming calls)

Other CODECS:

- **G711aLaw** (also called PCMA)
- G729r8