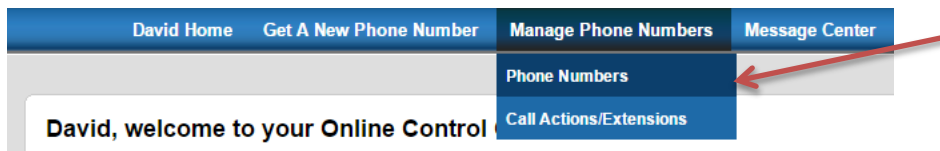


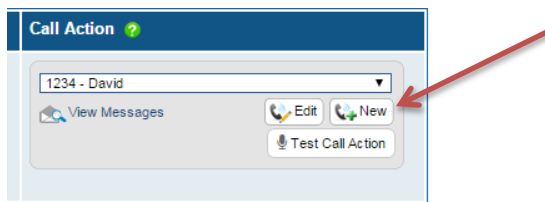
SIP Instructions

Enter a [SIP Ring-To address](#) on the TollFreeForwarding.com website

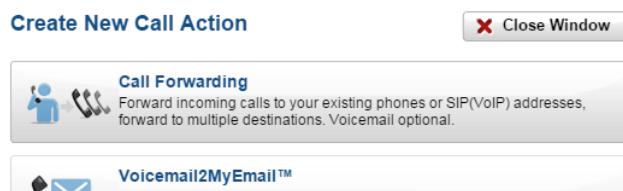
1. From the Home page of your Online Control Center, choose Manage Phone Numbers > Phone Numbers



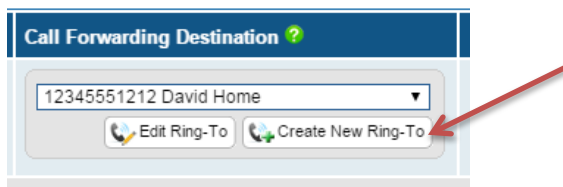
2. Choose "New" under Call Actions



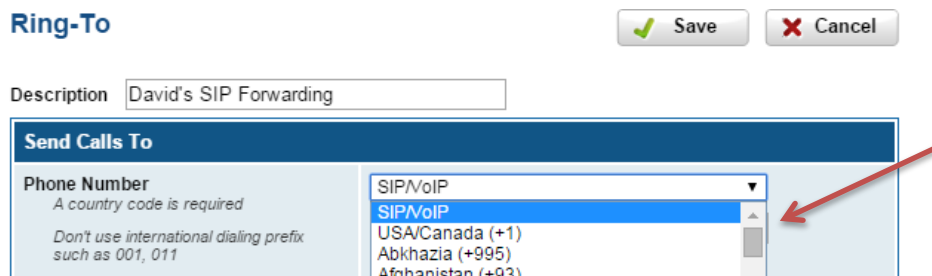
3. Next, select "Call Forwarding"



4. Then "Create New Ring-To"



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



Ring-To Save Cancel

Description

Send Calls To

Phone Number ▼

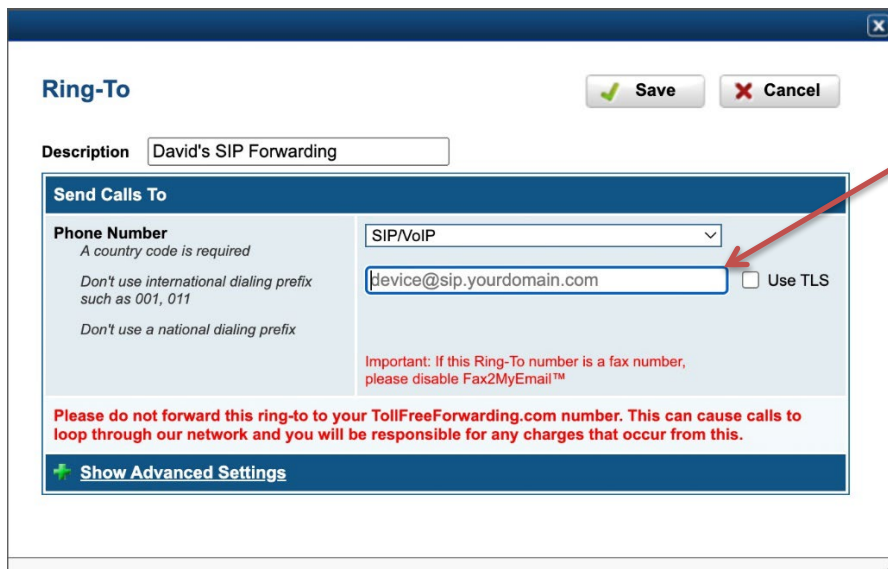
A country code is required

Don't use international dialing prefix such as 001, 011

Don't use a national dialing prefix

SIP/VoIP
USA/Canada (+1)
Abkhazia (+995)
Afghanistan (+93)

- Enter the SIP address that will be receiving your calls



Ring-To Save Cancel

Description

Send Calls To

Phone Number ▼

A country code is required

Don't use international dialing prefix such as 001, 011

Don't use a national dialing prefix

Use TLS

Important: If this Ring-To number is a fax number, please disable Fax2MyEmail™

Please do not forward this ring-to to your TollFreeForwarding.com number. This can cause calls to loop through our network and you will be responsible for any charges that occur from this.

[+ Show Advanced Settings](#)

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

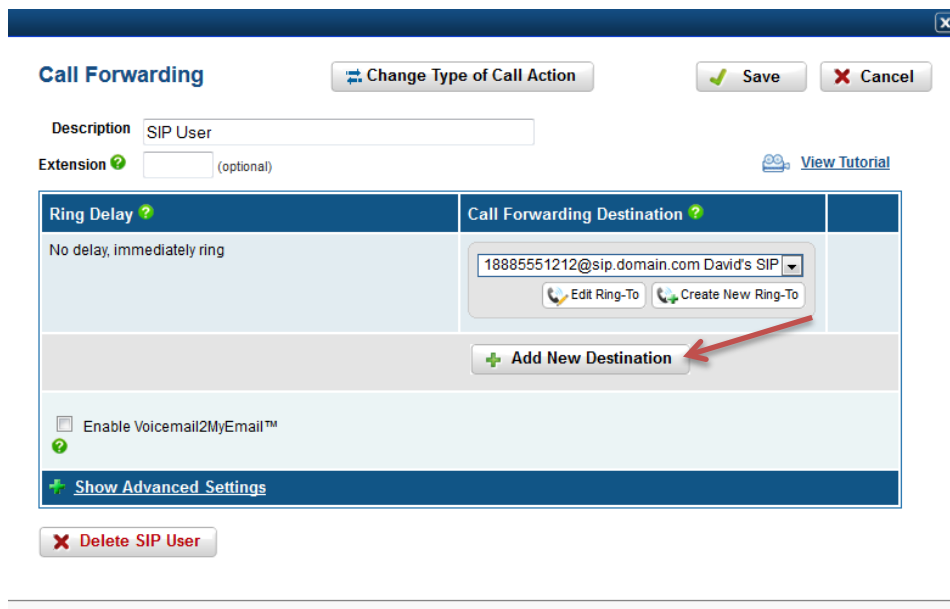
Ring-To Notes:

- 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.

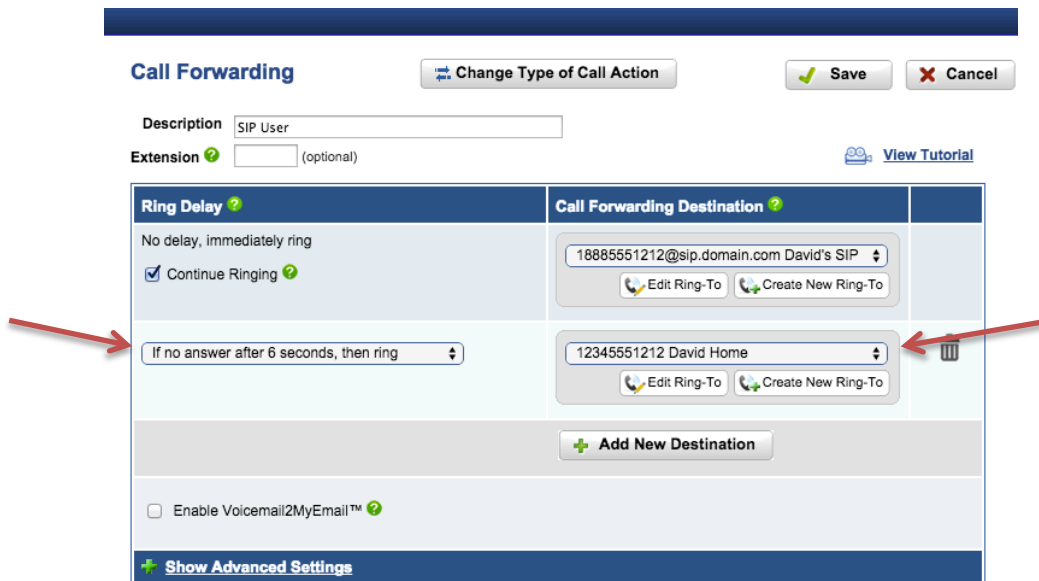
- Within the call forwarding action, click “Add New Destination”.
- Choose the Ring-To number from the drop down list, or click “Create New Ring-To” and enter the new number to forward to.



The screenshot shows the 'Call Forwarding' configuration page. At the top, there are buttons for 'Change Type of Call Action', 'Save', and 'Cancel'. The 'Description' field contains 'SIP User' and the 'Extension' field is empty with '(optional)' next to it. Below this is a table with two columns: 'Ring Delay' and 'Call Forwarding Destination'. The 'Ring Delay' is set to 'No delay, immediately ring'. The 'Call Forwarding Destination' dropdown menu is open, showing a list of destinations including '18885551212@sip.domain.com David's SIP'. Below the dropdown are buttons for 'Edit Ring-To' and 'Create New Ring-To'. A red arrow points to the '+ Add New Destination' button. At the bottom of the table, there is a checkbox for 'Enable Voicemail2MyEmail™' and a '+ Show Advanced Settings' button. Below the table is a 'Delete SIP User' button.

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 Change Type of Call Action Save Cancel

Description

Extension (optional) [View Tutorial](#)

Ring Delay	Call Forwarding Destination
No delay, immediately ring <input checked="" type="checkbox"/> Continue Ringing	18885551212@sip.domain.com David's SIP Edit Ring-To Create New Ring-To
If no answer after 6 seconds, then ring	12345551212 David Home Edit Ring-To Create New Ring-To

[+ Add New Destination](#)

Enable Voicemail2MyEmail™

[+ 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	52.144.10.100
Los Angeles Secondary Datacenter	52.144.11.100
Global (TLS Only)	52.144.10.95
Global (TLS Only)	52.144.11.95

RTP Media IP Addresses				
Note: Most systems do not need to be configured for these specific RTP Media IP addresses				
	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 **ONLY** supported by **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**