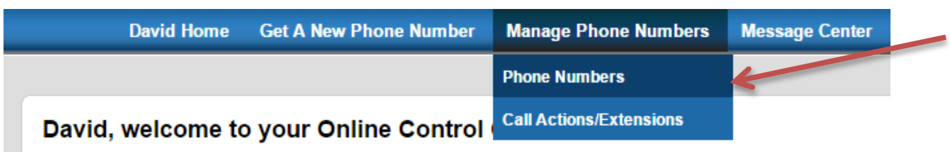
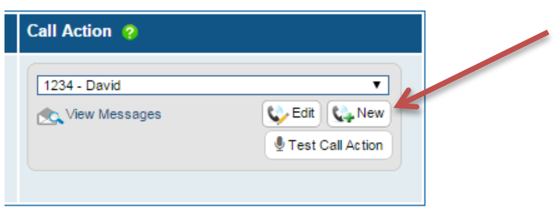
SIP Instructions

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

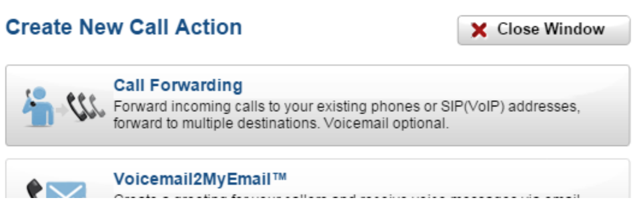
1. From the Home page of your Online Control Center, choose Manage Phone Numbers > Phone Numbers



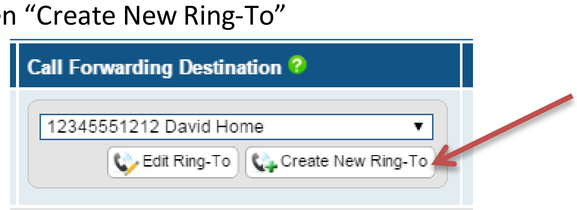
2. Choose “New” under Call Actions



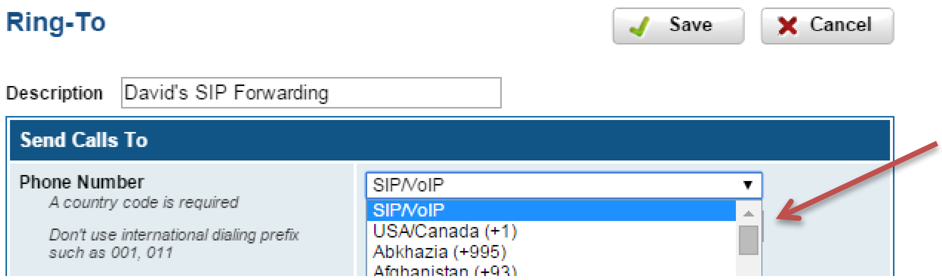
3. Next, select “Call Forwarding”



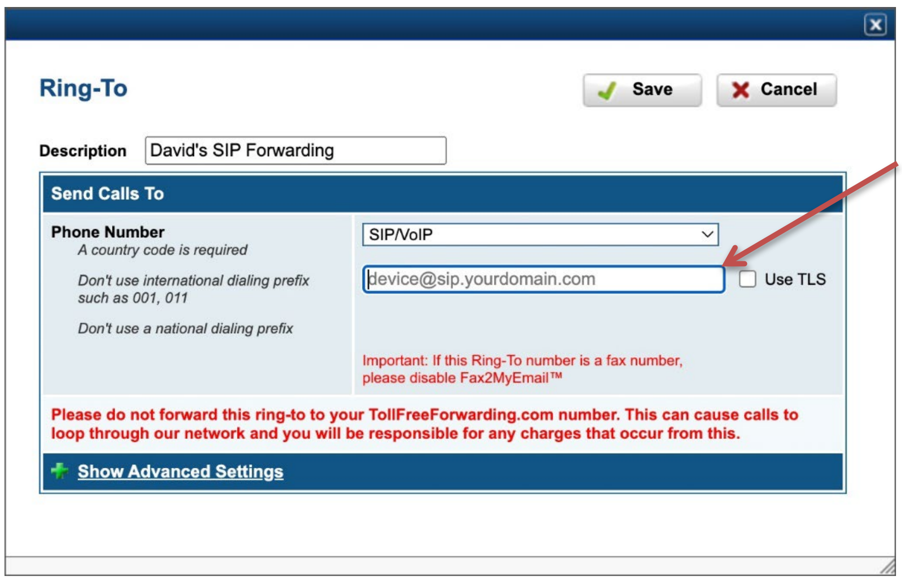
4. Then “Create New Ring-To”



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



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



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:

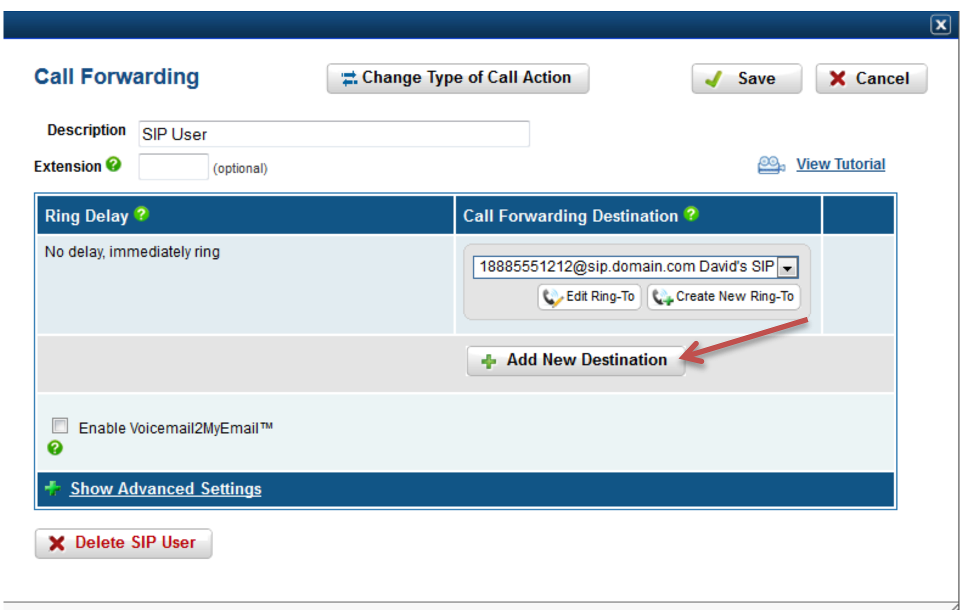
- The IP address may have fewer numbers between the periods, but it always has four sets of numbers separated by three periods: @1.2.3.4.
- Unless you are very familiar with DNS, most of our customers find it easier to use the IP address of their SIP server instead of DNS.
- Make certain that the user (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)

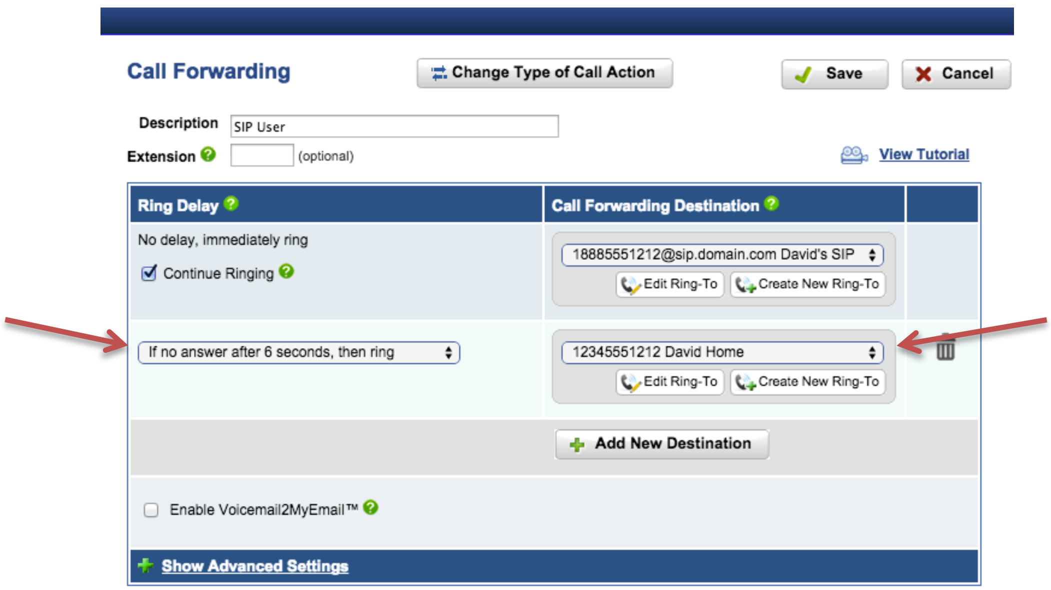
Note: Multiple ring-to destinations may be set up to include a mix of SIP addresses and phone numbers.

- 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.



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.



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**