6902		**Asterisk 192.168.71.51**			**911122223333**	**RTP State**	
:---->	INVITE 6120@192.168.71.51						
	TRYING	<---:					
	OK 192.168.71.51:11148	<---:				sendrecv 192.168.71.51 port 11148	
:---->	ACK 192.168.20.191:1044					sendrecv 192.168.20.191 port 1044	
							Call established between Cisco 6902 and Asterisk
		:---->	INVITE 911122223333@192.168.10.10			sendrecv 192.168.71.51 port 17690	
			TRYING	<---:			
			RINGING 192.168.10.10:24706	<---:			
			OK 192.168.10.10:24706	<---:		sendrecv 192.168.10.10 port 24706	
		:---->	ACK				Call established between Asterisk and 911122223333

Two legs of the call are bridged within Asterisk.
Asterisk reinvites the two legs of the call so that RTP path by-passes Asterisk and is sent between Party A and Party B.

	INVITE 6902@192.168.10.10	<---:				sendrecv 192.168.10.10 port 24706	
:---->	TRYING						
:---->	OK					sendrecv 192.168.20.191 port 1044	
	ACK	<---:					
		:---->	INVITE 911122223333@192.168.10.10			sendrecv 192.168.20.191 port 1044	
			TRYING	<---:			
			OK	<---:		sendrecv 192.168.10.10 port 24706	
		:---->	ACK				

The RTP path is now re-INVITED between 192.168.20.191 (Cisco 6902) and 192.168.10.10 (Cisco Call Manager). Note that Cisco Call Manager will probably re-invite the call to remove Call Manager from the RTP path, meaning that the RTP of the call would be between the Cisco handset and Cisco gateway.

The call is now placed on hold on the Cisco handset.

:---->	INVITE 6120@192.168.71.51				inactive 0.0.0.0 port 1044
	TRYING	<---:			
	OK	<---:			inactive 192.168.10.10 port 24706
:---->	ACK				
		:---->	INVITE 911122223333@192.168.10.10		sendrecv 192.168.71.51 port 17690
			TRYING	<---:	
			OK	<---:	sendrecv 192.168.10.10 port 24706
		:---->	ACK		

The 6902 channel is inactive. The 911122223333 channel has been re-INVITED back to Asterisk.
Music on hold starts on channel 911122223333 - 192.168.10.10 port 24706

Cisco now sends an INVITE with no SDP.

:---->	INVITE 6120@192.168.71.51		
	TRYING	<---:	
	OK	<---:	sendrecv 192.168.10.10 port 24706
:---->	ACK		sendonly 192.168.10.10 port 4000

Cisco acknowledges the call with sendonly. It now plays music on hold to 192.168.10.10 port 24706

Asterisk re-INVITES 902084019108 channel back. Music on hold stops because Asterisk thinks that the call
has been taken off hold (it has not yet received the sendonly SDP from Cisco)

:---->	INVITE 911122223333@192.168.10.10		sendrecv 192.168.71.51 port 17690
	TRYING	<---:	
	OK	<---:	sendrecv 192.168.10.10 port 24706
:---->	ACK		

Once the 911122223333 chanel is re-INVITED back, Asterisk has received the ACK with sendonly from Cisco. It plays music on hold to 192.168.10.10 port 24706.
When Asterisk receives sendonly from Cisco it should not send an RTP stream to 192.168.10.10 port 24706.

At this point Party B can hear Asterisk music on hold and Cisco music on hold because both RTP streams are pointing to 192.168.10.10 port 24706.

Party A takes the call off hold.
Cisco first places the RTP in inactive state between stopping music on hold and re-connecting the call.

:---->	INVITE 6120@192.168.71.51	inactive 0.0.0.0 port 4000

	TRYING	<---:			
	OK	<---:			inactive 192.168.10.10 port 24706
:---->	ACK				

Asterisk music on hold stops and starts on channel 911122223333

Cisco now reconnects the call.

:---->	INVITE 6120@192.168.71.51				
	TRYING	<---:			
	OK	<---:			sendrecv 192.168.10.10 port 24706
:---->	ACK				sendrecv 192.168.20.191 port 1046

Asterisk stops generating music on hold for channel 911122223333

Party B is re-INVITED

		:---->	INVITE 911122223333@192.168.10.10		sendrecv 192.168.20.191 port 1046
			TRYING	<---:	
			OK	<---:	sendrecv 192.168.10.10 port 24706
		:---->	ACK		

Now Party A and Party B are reconnected. The RTP stream has been re-INVITED so that Asterisk is not in the middle.

6902 hangs up the call

:---->	BYE				
	OK	<---:			

Channel 911122223333 is re-INVITED back to Asterisk to hangup the call.

		:---->	INVITE 911122223333@192.168.10.10		sendrecv 192.168.71.51 port 17690
			TRYING	<---:	
			OK	<---:	sendrecv 192.168.10.10 port 24706
		:---->	ACK		
		:---->	BYE		
			OK	<---:	