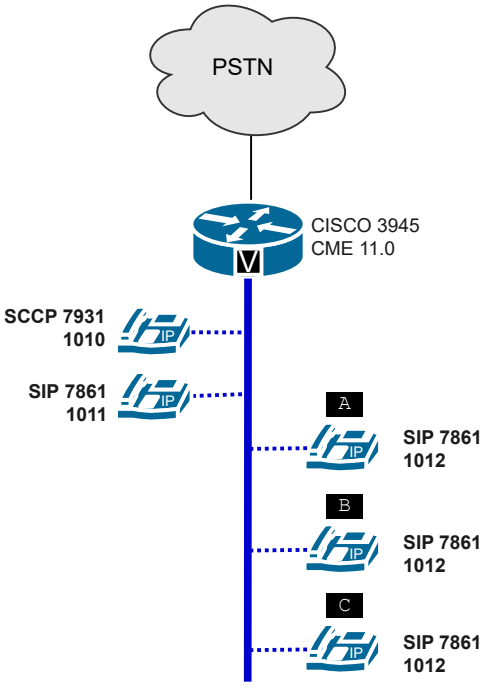
Problem with call waiting in CME SIP phones

1. Scenario and configuration

```
ephone-dn 1
  number 1010
!
ephone 1
  device-security-mode none
  mac-address D824.BDBB.0000
  codec g729r8
  type 7931
  button 1:1
!
voice register dn 1
  number 1011
  allow watch
  shared-line
  mwi
!
voice register dn 2
  number 1012
  allow watch
  shared-line
  mwi
!
voice register pool 1
  busy-trigger-per-button 2
  id mac 7035.0000.1011
  type 7861
  number 1 dn 1
  dtmf-relay rtp-nte
  username 1011 password 123456
  codec g711ulaw
!
voice register pool 2
  busy-trigger-per-button 2
  id mac 7035.0001.1011
  type 7861
  number 1 dn 2
  dtmf-relay rtp-nte
  username A002 password 123456
  codec g711ulaw
!
voice register pool 3
  busy-trigger-per-button 2
  id mac 7035.0002.1012
  type 7861
  number 1 dn 2
  dtmf-relay rtp-nte
  username B003 password 123456
  codec g711ulaw
!
voice register pool 4
  busy-trigger-per-button 2
  id mac 7035.0003.1012
  type 7861
  number 1 dn 2
  dtmf-relay rtp-nte
  username C004 password 123456
  codec g711ulaw
!
```




```
voice register global
 mode cme
 source-address ... port 5060
 timeouts interdigit 2
 authenticate register
 authenticate realm all
 tftp-path flash:
 file text
!
voice service voip
 no ip address trusted authenticate
 callmonitor
 no callmonitor trace
 rtp-port range 20000 30000
 dtmf-interworking rtp-nte
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12 advertise-
only
 no supplementary-service sip moved-
temporarily
 no supplementary-service sip refer
 no supplementary-service sip handle-
replaces
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy
0 hs-redundancy 0 fallback none
 modem passthrough nse payload-type 101
codec g711ulaw redundancy maximum-
sessions 1
 h323
 sip
  bind control source-interface ...
  bind media source-interface ...
  registrar server
  sip-profiles 1
  no call service stop
!
telephony-service
 no auto-reg-ephone
 ip source-address ... port 2000
 calling-number local
 service phone paramEdibility 1
 service directed-pickup gpickup
 service dss
 timeouts interdigit 2
 cnf-file location flash:
 cnf-file perphone
 load 7931 SCCP31.9-4-2SR2-2S
 max-conferences 8 gain -6
 call-park system application
 call-forward pattern .T
 dn-webedit
 time-webedit
 transfer-system full-consult
 transfer-pattern .T
 secondary-dialtone 0
!
```

2. Case 1: extension 1010 (SCCP) calls 1012 (SIP)

When 1010 calls to 1012:

2.1 - All devices (A, B and C) get ring.
2.2 - **A** answers.
2.3 - But the call is for **C**.
2.4 - So, **A** put the call on hold.
2.5 - In the all phones (A, B and C), the line led will blink.
2.6 - So, C pick up the call and talk to calling number (1010).
2.7 - That is, it works perfectly here.

Obs.: It's OK too if the call is originated from PSTN.

3. Case 2: extension 1011 (SIP) calls 1012 (SIP)

When 1011 calls to 1012:

2.1 - All devices (A, B and C) get ring.
2.2 - **A** answers.
2.3 - But the call is for **C**.
2.4 - So, **A** put the call on hold.
2.5 - In the all phones (A, B and C), the line led will blink.
2.6 - So, **C** (TRY) pick up the call and the confusion begins!
2.7 - When **C** presses the line button to pick up the call,
he makes a new call to the calling number (1011).
2.8 - Then the caller who is on hold receives a new call...
The mess is ready!