-05 draft submitted

- All outstanding issues dealt with some exceptions
  - Exceptions covered in a later slide

- Paul K submitted some comments
  - I will make all the suggested wording changes
  - I will fix the registration text as suggested
  - I will change the title of 5.2.2
  - I will fix two stage dial around text to specify use of the front door number
  - I will reword the phone number text to just say it has to conform to the RFC (meaning "+" “country code” “local number” with no decoration)
  - All AoRs will be phone numbers (no sip uris without a phone number)
  - Should we split config into subscriber/non-subscriber?
  - Single Stage Dial Around signaling (next slide)
Single Stage Dialaround

• All calls go to the default provider initially, just like any other SIP provider
• The signaling need only tell the default provider that single stage dial around is requested, and who the provider is
• The original idea is that the To: is \texttt{sip:<TN>@dialaroundProvider; user=phone}
  • A non single stage dial around To: is usually \texttt{sip:<TN>@defaultProvider;user=phone}
  • A tel uri could be used instead
• At one point we said use a Route header that specified the dial around provider, and ignore the domain in the sip URI
• What do we want to do?
  • Suggest using the original idea of \texttt{sip:<TN>@dialaroundProvider}
Not done

- IPv6 required
  - Some restrictions may be okay to simplify implementations
- Contact sync as is
  - jCard could be xCard but that really does seem backwards for a new spec.
- WebRTC compatibility still there
  - VP8 is optional
  - OPUS is required
  - WebRTC compatibility is in the charter.
  - A gateway to SIP would be required to use it.
- MWI support is required. Paul suggests it only needs to be required if the Provider offers voicemail
  - But everyone does VM, so is that actually necessary?
  - Have to add an MWI server URI to the config
Next Steps

• Rev to pick up Paul’s comments.
  • This is ready to go now, pending what we decide today

• Start Last Call