Connecting to the R&E Telepresence Exchange

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Part 1

CISCO TELEPRESENCE IN THE R&E TELEPRESENCE EXCHANGE
The R&E exchange community: traditional Cisco TP rooms

- Currently almost 200 rooms connected, many more ‘out there’
- About 80 institutions, most in US
- Single-screen and multi-screen
What is the R&E TP Exchange?

- Begun 2009, the central infrastructure that enables highly-functional, scalable, interconnection of many local, state/regional, and international telepresence systems
- Originally Cisco Telepresence, but not intended to be limited to Cisco only—now also:
  - Interoperability gateway, standards-based interconnectivity to other SIP & H.323 devices
What *is* that 'central infrastructure'?

- "SBC" Session border controller—performs call-admission, number analysis, call-routing, trunking
- Periodically monitors state of trunks via SIP "OPTIONS ping", a kind of SIP 'hello'
- Telepresence server blade, H.323 interop services
- Monitor quality of connections via Cisco IPSLA
  - Loss
  - Jitter
  - Latency
Central infrastructure 2

- Redundantly routed via NLR/Internet2 backbones
- Located in Kansas City in the NLR POP
- SIP trunk to each remote site or exchange
- Trunks to other exchanges (R&E + commercial)
- Interconnected with Internet2 backbone for reachability to Internet2 members
  - Exchanges limited routes with I2 for Telepresence
- Multipoint services (CTMS)
Central services
what do we do for you?

- Coordinate testing on turnup
- Coordinate R&E telepresence site directory
- Maintain mailing list for news, alerts, q&a, and website for FAQ & other information
- Represent community to vendors, providers
  - Enroll sites with commercial providers
- With community, help set standards
- *Not* end-user support, hardware support
Minimally:

- Codec/screen/IP phone user interface
- Cisco Call Manager (CM) for managing (up to many) endpoints, signaling, terminate trunk, call-routing, managing software, reporting, etc etc

CM function could be shared with other institutions
Optionally:

- Redundancy, e.g. CM cluster
- Local multipoint switch
- Local interop options to other SIP or H.323 devices
- firewall/border device(s)
- Recording
Functional options for endsites

- NAT
- CTS-manager for scheduling, integration w/ Outlook, can push calendar to phones
- Encryption
- PSTN gateway
Requirements of endsites

1. Routed IP address(es) for Call Manager
2. Routed IP address(es) for codecs
3. E.164 “phone number” for codec: our standard is an ‘internationalized’ E.164 number correct for your locality. In North America, 11 digits: $1+ (\text{area code})+(\text{exchange})+(\text{local part})$

   For example (US) 1-919-123-4567 or (China) 86-1-21-12345

   CM understands international dialing

   Doesn’t need to be switchable; PSTN connection is optional

4. That number is your ‘dialing number’ outside, and you must answer when other sites call you with that.
Sample R&E TelePresence Components & Layout

Minimal end-site configuration
CUCM & codec

More complex end-site configuration
More CUCMs, more codecs

Optional end-site components
- firewall
- CTMS (multipoint Switch)
- CUBE-Ent (security, Signal demarc)
- CT-MAN (scheduling, Management)
- MXE/MSE (interop)
- CUV (interop)
- PSTN interop
What's needed to connect?

20,327’ view:
You have a codec & call manager. To connect to & use the R&E exchange, you need:
1. IP reachability: A functional routed (layer3) connection that can reach the exchange
2. A SIP trunk to the exchange
3. A valid E.164 (phone) number & dial plan
Details #1: routed connection

- Traffic must be able to flow freely
  - All protocols are documented well
  - SIP signaling Call Manager → SBE 216.24.184.130
  - Media flows codec → DBE 216.24.184.131
  - Signaling on SIP port 5060/5061, media UDP RTP 16-32K

- Leverage existing high-performance networks
  - Only ~5Mbs/screen, no special circuits needed

- Traffic must be loss-free, low-latency, low-jitter
routed connection—what can go wrong?

- Firewall problems, for example letting signaling AND media through, or not getting enough SIP state. Sometimes the fix is to insert a CUBE (proxy).
- NAT: ‘nuff said?
- Have to adjust routing for Internet2 members to get to NLR POP where R&E exchange is located.
- Loss, latency, jitter: jitter & latency issues are rare, but loss sometimes needs to be fixed with QoS. Bandwidth issues are very rare in our networks.
Details #2: SIP trunk

Persistent SIP adjacency is created between CM and SBC by creating a SIP *trunk*

- Uses IP addresses of each end
- Since the trunk is stateless, the SBC periodically polls the CM over the trunk with an OPTIONS type of SIP packet to see if it answers. This hello-like interaction is called an ‘options ping’ though there’s no ICMP involved. The SBC can mark the adjacency as *online* or *offline* based on response.
Creating the SIP trunk (in CM)

<table>
<thead>
<tr>
<th>Destination Address is an SRV</th>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="216.24.184.130" alt="Marked IP Address" /></td>
<td><img src="216.24.184.130" alt="Marked IP Address" /></td>
<td><img src="216.24.184.130" alt="Marked IP Address" /></td>
<td>5060</td>
</tr>
</tbody>
</table>

- MTP Preferred Originating Codec: 711ulaw
- Presence Group: Standard Presence group
- SIP Trunk Security Profile: Non Secure SIP Trunk Profile
- Rerouting Calling Search Space: < None >
- Out-Of-Dialog Refer Calling Search Space: < None >
- SUBSCRIBE Calling Search Space: < None >
- SIP Profile: Standard SIP Profile
- DTMF Signaling Method: No Preference
SIP trunk—what could go wrong?

- If protocol path is opened correctly, this should work fine and almost always does.
- For (us) data people, SIP is generally a foreign language: how to decipher what exactly was wrong, or missing, in the negotiation?
- This is where we may see configuration issues with other parts of the CM or codecs, for example, wrong protocol or bandwidth settings.
Details #3: number & dial plan

- End site designates a valid E.164 number for each device (see our *standard* earlier)
- Number is programmed into the device via CM, associated w/ IP of known registered device
- Phone & codec are associated by virtue of same E.164
- .... more
Details #3: Dial plan (p.2)

- CM may have various trunks, dial-plan routes destination number (patterns) to trunks
  - Uses longest-match (most-specific) pattern
  - Knows all ‘local’ devices automatically
  - Generally punts everything else to exchange
  - So it’s possible to have a single dial-pattern: “@”

- CM may perform number manipulation on incoming or outgoing numbers
Dial plan—what could go wrong?

- One of the most frequent problems is that the CM uses a short version of the long phone numbers locally, and doesn’t recognize the full number when it comes in, refusing the call.
- Sometimes the CM doesn’t format the reporting of the *outgoing* number correctly.
- Unnecessarily complex dial plans
- User confusion with TP, PSTN, local/LD prefix
- Wrong numbers—directory helps!
How does TP connection work?
‘above’ and ‘below’ the covers…

- Codec & phone register to Call Manager (CM)
- CM loads image & config (including directory, calendar) to codec & phone
- User dials (manually or via directory) number
- Codec signals call to CM (SIP)
- CM compares with dial plan, signals call to SBC
- …more
How does it work? #2

- SBC receives signaled call from CM
- SBC compares with dial plan, routes call to appropriate end-site trunk (incl interop sites)
- Remote CM receives signal, analyzes called number & call requirements if it wants to answer
- Remote CM signals orig CM (via SBC) that call is ok, state ‘active’, start to send media (via SBC)
- UDP Media begins to flow codec to SBC to codec
When >1 system is in call, uses a ‘multipoint switch’ (CTMS)

- Just another SBC trunk, chosen by SBC’s dial-plan
- No transcoding is necessary if all Cisco
- Up to 48 screens at once, expanding to 90
- Screen-switching, or site-switching
- Supports encryption, blocking, listing, dial-out
- All callers call the same E.164, CTMS joins them together

Looks like a normal call
Inter-exchange calling uses the same fundamentals: IP connect, SIP trunk, dial plan

- Usually need to create a new physical connection
- Single or redundant trunks between exchanges
- Dial plan selects correct trunk
- Commercial exchanges don’t allow p2p dialing, only connect via their multipoint switches
  - Pro: Only one number for us to call for each
  - Cons: no p2p, no interop
Until now, requires a transcoding box, (see Ben’s part of this presentation)

_Telepresence Interoperability Protocol_

Starting summer 2011, new code allows direct p2p calls with endpoints that support “H.264 baseline” standards (8.6 in CM, 1.7.4 codec)
Who? Where?

There is an internal directory in the phone

Populated from CM

Same for all phones registered to that CM

Can be created (CSV) & uploaded to CM

Can have 100s of numbers
Directory—globally

- How do you find out what’s out there, and where?
- How do you find who to talk with about it?
- What’s its ‘phone number’?
- As user, how do you control visibility?
- How do you do these with PSTN or web today?

- R&E exchange directory
- North Carolina State University TP directory
- Cisco TP directory
- Commercial-provider directories
- Nothing global…
- No mechanism for auto-listing…
Phone ‘favorites’

- Configured from CM
- Appears on IP phone
- Different for each phone
Telepresence room as a resource

- Two parts:
  - See availability
  - Commit availability

- How large a view is appropriate?
  - Can you schedule someone else’s resources?
  - Should you be able to? (A&A issue)
  - Should you be able to see if/when they’re available?
Calendaring p.2—*intra-enterprise*

- Can schedule *your* devices (codecs, CTMS) as resources (mail, web) with *CTS-man* appliance, integrating resource reservation nicely with phone itself
- *CTS-man* can connect to groupware calendar/resource-mgt app (e.g. Outlook), or other apps via API
- *CTS-man* pushes calendar to phone, ‘one button’ call
- No good *inter*-enterprise solution today (API?)
- No ‘open’ way to connect calendar to phone/CM
For more information

- Nlr.net pages on Telepresence, including FAQ & map of connected sites
- Noc.nlr.net > Documentation > Telepresence for information on many aspects of connection and maintenance of your connection, including:
  - Dial plan information & instructions
  - List of connected endsites
  - How-tos, configuration guides
  - GRNOC router proxy gives you visibility