Avaya Demystified
by Walt Medak

Q: We are in the process of cutting over from our trusty old Avaya system to that "other brand" of VoIP phone systems. We are using both systems currently, with a tie line set up between them. We just installed some SIP trunks in the "other" system, and want to start routing some of our outbound calls from the Avaya phones across the tie line and out the SIP trunks. The trick is, my boss wants me to set this up based not on the number dialed, but on the Class of Restriction of the phone. I know how I could do it based on the phone number using ARS Analysis and route patterns, but I don't know how I can do it with the COR. What do I need to do?

A: There is a way to route calls based on a setting that's in the COR, assuming you have a couple options enabled in your system. This way uses time of day routing and ARS partitioning. You will need to create new route patterns to send calls over the tie line. How many you will need to create will depend on how complicated your existing routing is for local calls, long distance calls, toll free, 911, etc...
Once you have all the new route patterns set up, you will need to modify the partition routing table. I would use partition group number (PGN) 1 for the phones that will remain routing over the Avaya trunks for now, and partition group number (PGN) 2 for the phones that need to route to the SIP trunks. In the column for PGN 1, you would enter all of the existing route patterns for the calls to go over the Avaya trunks. Column PGN 2 would be for the corresponding route patterns for the SIP routing.
After completing the partition routing table, you will need to go through your entire ARS Analysis table and replace the route pattern number for each entry with the corresponding partition route. For example, if local calls used route pattern 2 and you put that in the partition routing table in row 3, you would replace the "2" with "p3".
Next, you would need to create new COR's for the phones that you want to route to the SIP trunks. The only difference between the old and new COR would be which Time Of Day chart to use. I would leave the existing phones as chart 1, and put the "SIP routing" phones in chart 2. Then, change the time of day routing table 2 to put the phones in PGN 2.
Finally, when you are ready to change a phone from routing over the Avaya trunks to the SIP trunks, you would just need to change that station's COR to one of the new ones you just created. This is a fairly complicated procedure to try to explain in a short forum like this. Please fell free to call if you have any questions.

Q: We have a handful of analog station ports programmed in our system that we have used over the years to provide dial tone and act like a POTS line for testing other phone systems. I needed to use one of those lines the other day to test an analog phone, but it didn't want to work for me. I can get dial tone, and receive calls on it, but I couldn't make an outbound call. I checked the programming on the station and it looked good as far as I could tell. It had the same COR and COS as other analog devices that work just fine. What am I missing here?

A: I had an idea what might be happening when you said you had been using the lines to imitate POTS lines in another phone system. I looked at the programming for those test lines and was able to verify my thoughts. They were programmed as hotline stations. If you look at the last page of the station form, you will see a section called "*HOT LINE DESTINATION*". If this section is programmed, the station will immediately dial whatever number is programmed via the abbreviated dial list. In your case, the stations were dialing a "*9*" as soon as they went off-hook, no doubt as part of the attempt to act like a POTS line for your testing. So, when you picked up the phone and tried to dial an extension, it wouldn't work because the system was expecting the digits for an outbound call. If you want those ports to act like regular stations again, all you need to do is remove the "*Abbreviated Dialing List Number*" and "*Dial Code*" information.

And as always, if questions please call 800-452-6477.