Why DO SIP CALLS STILL FAIL?

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Title: **Why do SIP calls still fail?**

This talk will go into the SIP VoIP issues I experienced in the field and examine what service providers should do to avoid them.

The list of issues is fairly long and I will share what I have learnt over the years working in Support and as a Solution Architect for a SIP “networking” equipment vendor and working with various SIP VoIP Service Providers and their customers.

The SIP VoIP services offered by the Service Providers are usually SIP trunking and Hosted PBX.

I will explain what I mean by failed SIP calls.

I will categorize / group various issue areas.

I will list many issues, but won’t be able to provide detailed examples to every issue, but I picked some, which hopefully will trigger your attention.

I will discuss / describe Best Practices and troubleshooting tools used by various service providers and by the Edgewater Networks Support team.

What is a failed SIP call?

For the purpose of the presentation I defined a failed SIP call as:

A SIP voice call that for some reason doesn't connect, has no or one way audio, drops for no apparent reason or has (very) bad audio quality.

Therefore - a failed call could be caused by various reasons including IP networks, users, admin/operators and the SIP protocol / interoperability errors.

Keep in mind - the user doesn't care what caused the failed call, (s)he only cares if the voice call works or doesn't, therefore your VoIP service offering needs to address any of these issues.

Note: Try a Google search for failed SIP calls and you will find reported issues throughout the years – and many times you see same issues "repeat" themselves over the years. Why is that?

Failed SIP Call -- Categories

I divided the possible issue areas into 4 Categories + % of issues we see

- IP Network (40%)
  - Router / FW / NAT/ Dynamic routing
  - Switches / VLAN / MTU settings
  - DNS
  - ALG / SIP proxy / SIP fixup
  - IP Packet impairments (loss, jitter, delay)
- SIP Protocol(20%)
  - Registration
  - Call Setup / Codecs
  - Mid Call
    - Drops / hold / transfer
  - BLF / Shared lines
  - DTMF
  - HMR
- Admin (20%)
  - IP Network (LAN / WAN)
  - VoIP network
  - Network owner?
  - SIP UA / US settings
    - PSTN GW grounding
  - SIP peering
- User (20%)
  - Dial delay
  - required digits to dial out
  - Star codes differ from old PBX / key-system
  - All these new buttons on my phone
  - Echo

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Note: Partial lists of errors in the different categories

Some Notes to the following Examples

- The examples from the different categories will show
  - Symptom the user sees
  - The issue(s) description itself
  - Possible Solution(s)
- Some examples might seem trivial to you, but you would be surprised how often we still run into it these issues in the field deployments even today.
- Some examples I chose are “corner” cases, I chose them since these are usually hard to solve and you might be able to use your newly applied Wireshark Ninja Skills ☺

IP Network Examples – “Classis” NAT and FW issue

- Symptom
  - Can’t receive calls
  - Can place a call, but one way audio, no inbound audio
- Issue
  - NAT is layer 3 to adjust TCP IP/Ports, but SIP SDP also requires to updated these IP addresses / ports
- Solution(s)
  - FAR-END NAT traversal between core SBC and phones – OTT Provider Model
    - Scaling issue – Core SBC / SS has to deal with phone registrations every 30 seconds
  - Use CPE router that understands SIP NAT
    - ESBC – various field proven options available (DON’T use “SIP fix-up” feature in your $50 router)
  - STUN / TURN / ICE
    - Often requires additional server at customer premise
    - Limited support on “basic routers / FW”

IP Network Examples – Dynamic Routing Issue

- Symptom
  - Everything seems to be up / ping-able and phones are registered
  - Sometimes call don't connect
  - Sometimes no/one-way audio
- Issue
  - Split or broken IP routes due to dynamic BGP routing settings and a broken link
  - Registrar, Media Server, GW are on different IP networks / routing issue
- Solution(s)
  - Get your IP troubleshooting tool-box out – this is hard to figure out
    - You will need to confirm "pings" from each component to every component on the network
    - You might have to "SIP debug" every SIP device to see where the traffic should go (IP/PORT)
  - Use Network management tool to capture traffic flows over long period of time

IP Network Examples – Least Cost Routing Issue

- Symptom
  - Everything seems to be up / ping-able and phones are registered
  - 1 out of 100 calls has really bad audio quality
- Issue
  - Least Cost Routing can mean lowest quality routing. Wholesale SIP provider might have 10 underlying carriers and routes can vary from call to call
- Solution(s)
  - Capture multiple “reference” trace-routes between involved devices
  - Use a VOIP Network management tool and capture traffic flows and MOS values over long period of time – set MOS trigger alerts and run trace-routes when it happens

IP Network Examples – MTU Mismatch

- Symptom
  - Everything seems to be up / ping-able and phones are registered
  - Basic SIP signaling work, but more complex SIP messages fail (e.g. BLF, Shared Calls)
- Issue
  - MTU mismatch, large packets are truncated instead of properly fragmented. Common with PPPOE or WAN-VLAN.
- Solution(s)
  - Capture SIP signaling -- this will show truncated SIP messages usually in one direction
  - Understand network setup in detail – usually this happens on the last mile link.
  - Adjust max MTU size accordingly to consider Ethernet frame overhead due to encapsulating of e.g. PPPOE frames.

Administrator / Operator Error Examples

- Symptom
  - I hear some noise on the line, the call is distorted (sometimes), the voice quality is horrible (sometimes)
- Issue
  - VoIP/PSTN gateway is not properly grounded
  - Packet loss can cause voice distortion (PLC = packet loss concealment, fill in missed packets)
  - Transcoding can cause voice distortion
- Solution
  - Sometimes its good to capture the RTP and play it back to narrow the possible issue
    - VoIP GW ground issue sounds very different than maybe a “robotic” sound
  - Use MOS score analysis tools – you might see large number of continuously lost packets

Administrator / Operator Error Examples --- DNS

- Symptoms
  - Phone doesn't register / calls don't work anymore
  - I used the "working " config template to configure the phone
  - I can't (or maybe can) ping the SIP server FQDN from my PC
- Issues
  - SIP Phone usually uses FDQN entry(ies) for the SIP server – therefore that address need to be properly resolved by the DNS server(s)
  - Often phones are deployed on separate voice VLAN, which might use different DNS server – sometimes internal DNS server that take longer to update with the public DNS server
  - FQDN IP address updates due to server move – can take a while to propagate update across whole network
  - SIP server might be set NOT to response to pings
- Solutions
  - Use ns-lookup to verify FQDN is resolved properly on the voice VLAN and then try ping
  - Capture traffic and see what fails (e.g. DNS lookup, SIP msg going to correct address – FDQN IP might be cached

SIP Protocol Examples --- Call drops

- Symptom
  - My call dropped in the middle I was talking, I might hear fast busy or just dead air
- Issue
  - Calls dropping at specific intervals (10 min, 30 minutes) due to Re-Invites messages that are not responded to – could be SIP config issue or IP network issue
  - Call dropping within the first minute – missing ACK in Invite
  - Could be caused by comfort noise algorithm (muted phone) and RTP silence detection
- Solution
  - SIP call flow capture to narrow down the issue(s)
  - Look for packet loss on network
  - Adjust SIP UA/US accordingly

User Examples – Abandoned Calls

- Symptom
  - The user (caller) hangs up since (s)he doesn't hear any ringing or dead air while on hold
- Issue
  - Could be the "dial-delay" settings on the phone are too long – easier to identify
  - Could be the 180 RINGING message is delayed or lost
    - Seen this on "larger" VoIP network, often PSTN gateway part of that call
  - Call on hold – silence suppression turned on and no MoH
- Solution
  - The basic SIP call flow might look correct, but it's a timing issue and the user hangs up
    - Requires SIP signaling capture – might have to run continuously for long period of time
  - Call hold – enable MoH, disable silence suppression, train the user

User Examples – Call Features behave differently for user

- Symptom
  - The call transfer fails – call just drops
  - Can't do 3 way calling as I used to
- Issue
  - On the old PBX pressing hook flash completed the transfer on the new VoIP phones it might hang up the call.
- Solution
  - User training (rule of 8x) -- user habits are well ingrained and they don't know they are doing it until you watch over their shoulder.
  - Try to mimic old PBX features on new SIP VoIP offering
  - A pcap alone might not help, you need to fully understand what the user wants to do and how (s)he does it

SIP Protocol Examples – BLF / Shared Calls / Sim Ring

- Symptom
  - User office admin doesn't get the correct BLF indications on phone console
  - Not all phones part of a Simultaneous Ring configuration ring
- Issue
  - BLF messages lost or only partial message received – large packet bursts 5x for status light updates
  - Key System Emulation / Simultaneous ring on 6 phones might be fine but starts failing with 6+ phones. Invites traffic burst overloads the connection or router buffers and acknowledgements gets lost.
- Solution
  - Might have to use TCP for Signaling – might cause "unwanted" retransmits
  - Find the bottle-neck, start on one end and follow the message hop by hop

SIP Protocol Examples – SIP Routing Error (Trunking Loop)

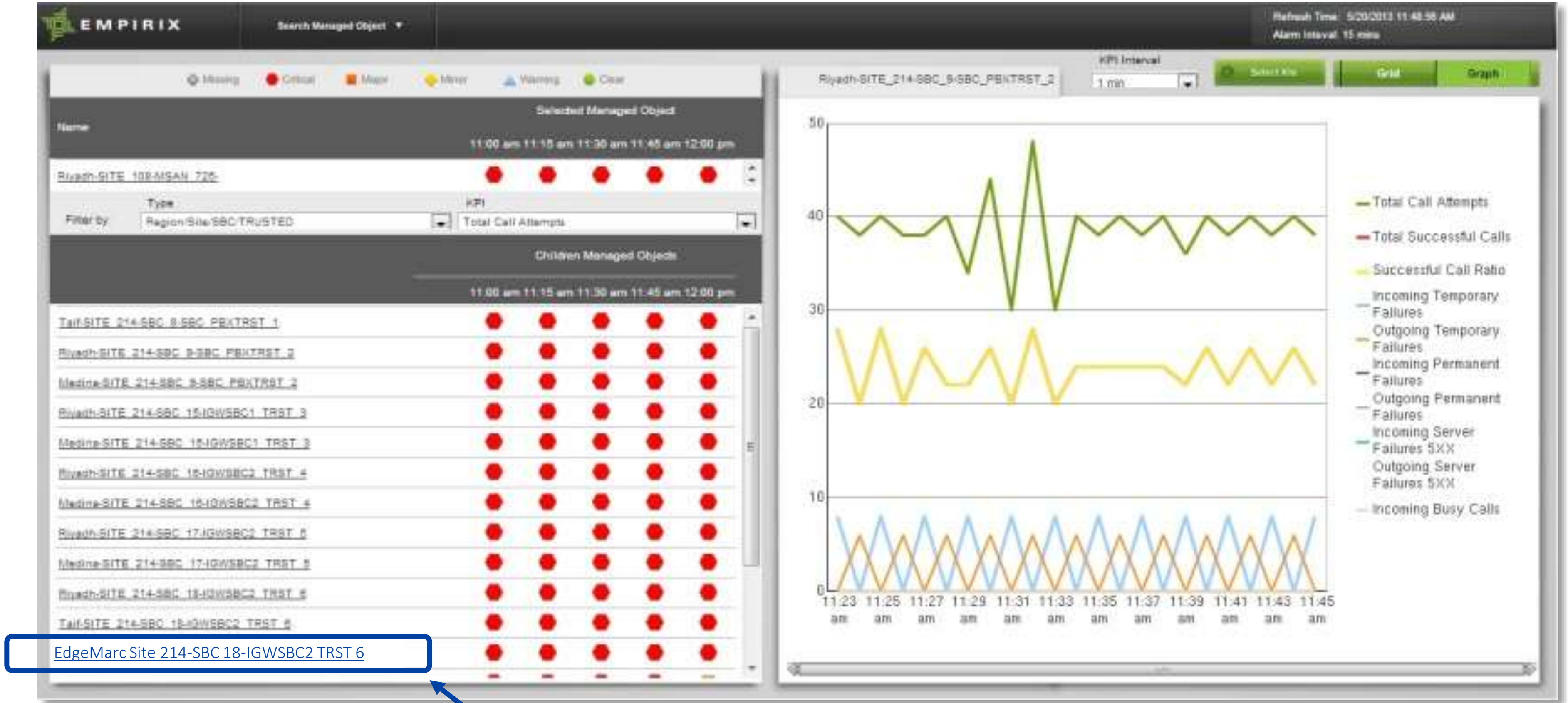
- Symptom
  - Inbound trunk call doesn’t connect – user not found
- Issue
  - Customer PBX call routing misconfigured – extension mapping
  - Call might get routed back out the SIP trunk (could even create a loop)
- Solution
  - Capture SIP call flow as close as possible to PBX
  - Make sure you use loop detection (RFC7332) on your VoIP network
  - Usually requires “co-operation” of PBX Admin – can be challenging

Some Best Pratices Items To Manage Your VoIP Network

- VoIP requires a “solid” IP network – make sure you can manage it
  - Use Data Network and VoIP Network Management Tools and make sure you can “link” them
- Automate whatever is possible
  - Device configuration (phones, SBC, SS, GW, …)
  - Trigger alerts based on various VoIP KPIs AND auto-collect required debug info
- The “VoIP Birth Certificate”
  - Take and store snapshot of your newly deployed and working VoIP network
  - Take periodic snapshots and compare to your previous ones
- Use a common data collection depository
  - Enable ongoing capture of your traffic (e.g. all SIP signaling)
  - Make sure you logs are time synced
- Try to have a single main monitoring screen based on the your role
  - E.g.: All tier 1 folks see the same screen

EVENT NOTIFICATION AND DRILL DOWN – 2 MGMT Systems linked

Add Event Notification in same workflow as IntelliSight



Hotlink to EdgeMarc status page in EdgeView

Use Case

1. TAC sees main dashboard which integrates high level alarm flow through to Empirix.
2. Integration of EdgeMarc alert into Intellisight
3. Interconnect Point at same level as Empirix probes.
4. For further troubleshooting into the call legs and scoring, a hotlink is provided to provide direct access to the EdgeView SCC.

There are many useful VoIP KPIs

Visualization can be used for many VoIP Services related
Key Performance Indicators (KPIs)

SIP Registrations related KPIs

- Number of Registered Clients
- Successful Registrations
- Failed Registrations
- Expired Registrations
- Number of registration attempts

Other SIP protocol related KPIs

- SIP Informational (1xx)
- SIP error codes (4xx, 5xx and 6xx)

Call Status related KPIs

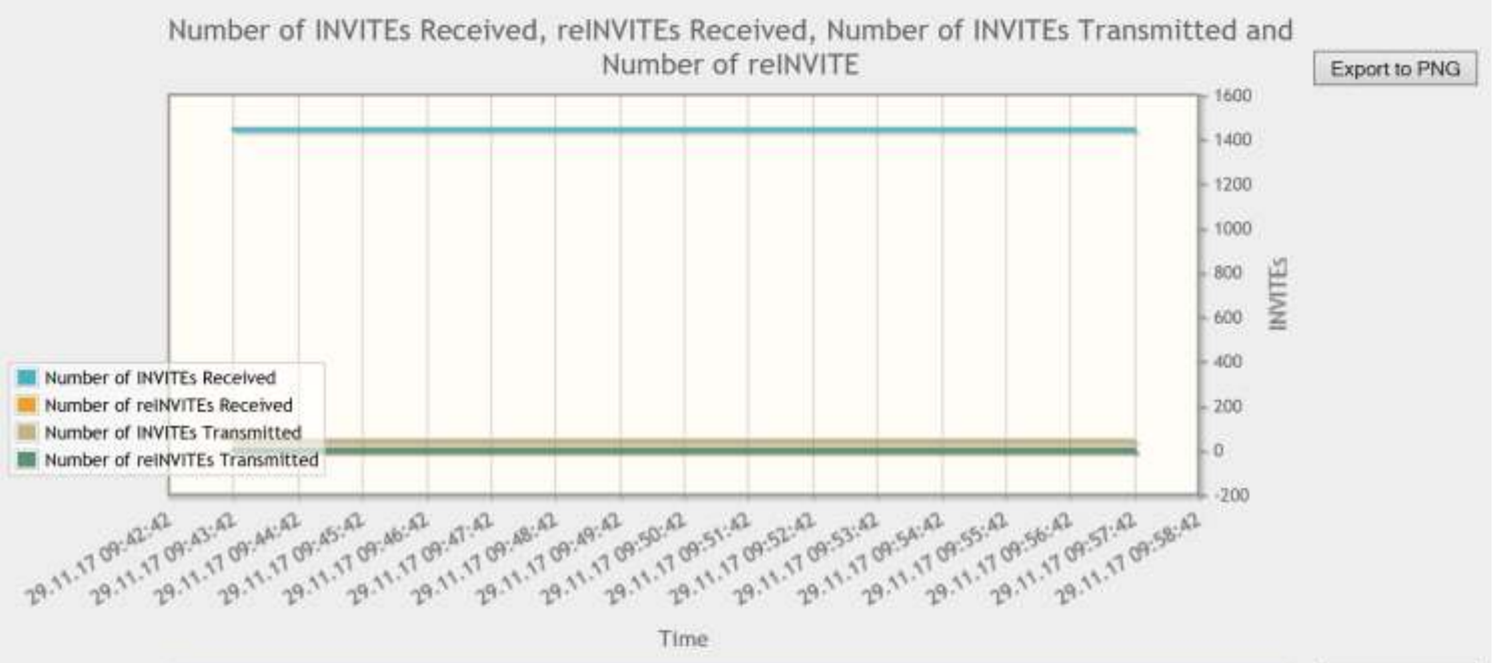
- Active connected calls
- Successful established calls
- Successful terminated calls
- Call setup delay
- Failed calls
- Abandoned calls
- Voice Codec used (G.711)
- VQM Statistics (lowest, average, highest MOS; ….)
- ……… (the list can go on) ……..

Other related KPIs

- Number of deployed SIP UAs (with SW version X; in geo-region Y)
- Bandwidth usage
- IP Network Packet Statistics
- Number of IP socket connections
- CPU usage
- RAM utilization
- ……… (the list can go on) ……..

VoIP KPI Example -- Invites

- # Invites Received
- # Invites Transmitted
- # Re-Invites Received
- # Re-invites Transmitted



Created DT	Number of INVITEs Received	Number of reINVITEs Received	Number of INVITEs Transmitted	Number of reINVITEs Transmitted
2017-11-29 09:57:42	1444	1	42	1

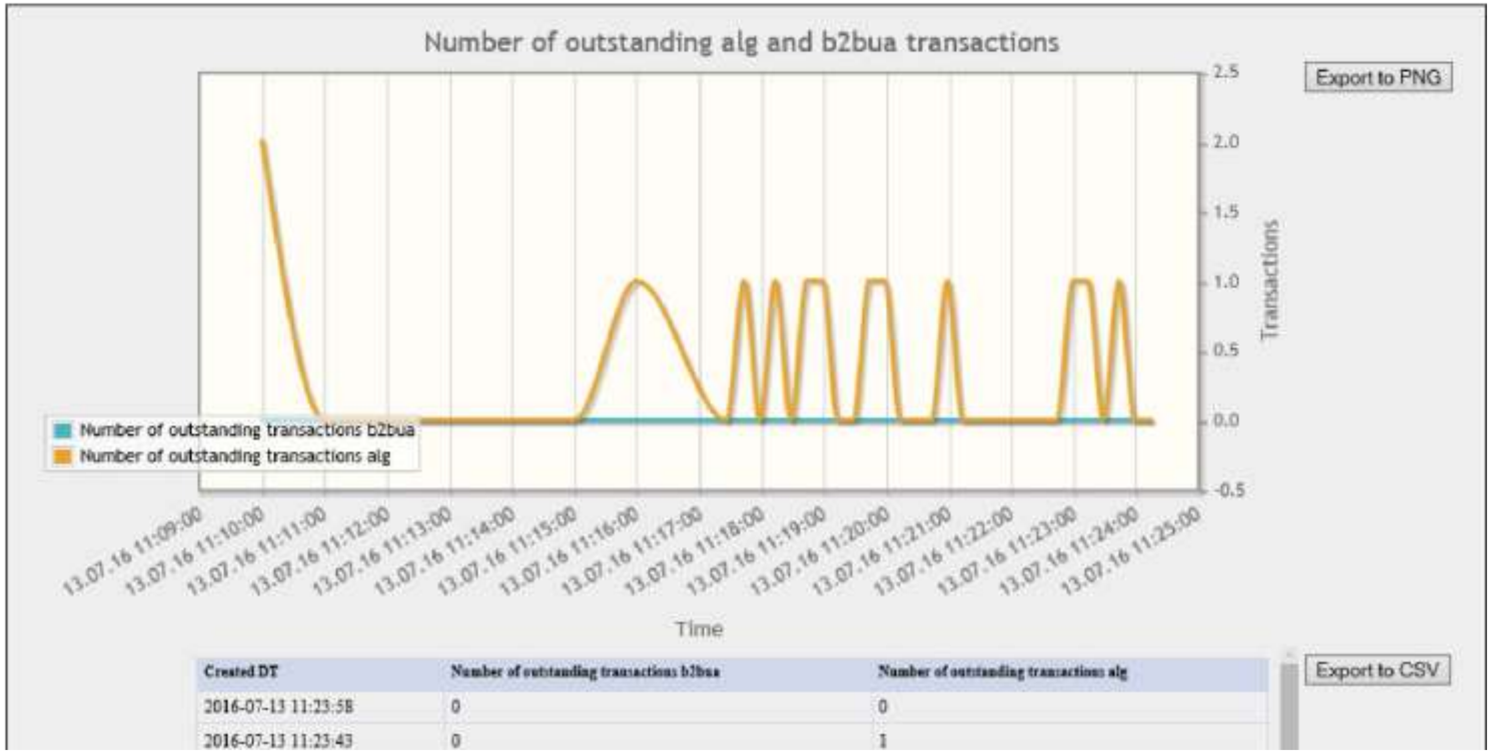
- Look for spikes and overall changes over time

VoIP KPI Example – SIP Transactions outstanding

- # outstanding ALG transactions
- # outstanding B2BUA transactions

- Look for spikes and overall changes over time
- Spikes might be expected / normal – varies between customer sites

Figure 8-17 Number of Outstanding ALG and B2BUA Transactions



Thank You

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