HIGH AVAILABILITY PBX

FROM SIERRA EXPERTS

Application Layer HA

What is High Availabilty PBX?

Generically, high availability (or HA) describes a system that is configured to ensure a prearranged level of operational service over a period of time. This is typically done using a 'clustering' approach, where multiple nodes are configured with failover. If a node or link in that cluster fails, service delivery can continue using the remaining node or nodes in system.

PBX is a prime candidate for high availability configuration because:

- In most implementations, failover is supported in the application protocol layer
- Voice traffic is often considered critical

Before understanding the implementation, understanding SIP trunking is prerequisite.

What is SIP Trunking?

A SIP Trunk is basically just a path that calls can take when signaling and routing using the SIP application layer protocol. There are several types of SIP trunks and many ways they can be set up, but all SIP compatible PBX implementations include the concept of trunking in one way or another. When interfacing with a service provider, that provider hands off calls using a SIP trunk. However, you can also set up your own SIP trunks between transmission devices, be it a switch, PBX, or B2BUA (Back to Back User Agent).

How can SIP Trunks Be Used to Build HA Solutions?

HA can be achieved in SIP by coupling PBXs with upstream soft switching infrastructure in a mesh configuration using SIP Trunks. In a mesh system, each node is connected to each other node, so in a mesh cluster of SIP devices, each device is connected to each other device using a SIP trunk. That way, if any given link (trunk) or node (device) fails, there is still a valid path between the source and the destination.

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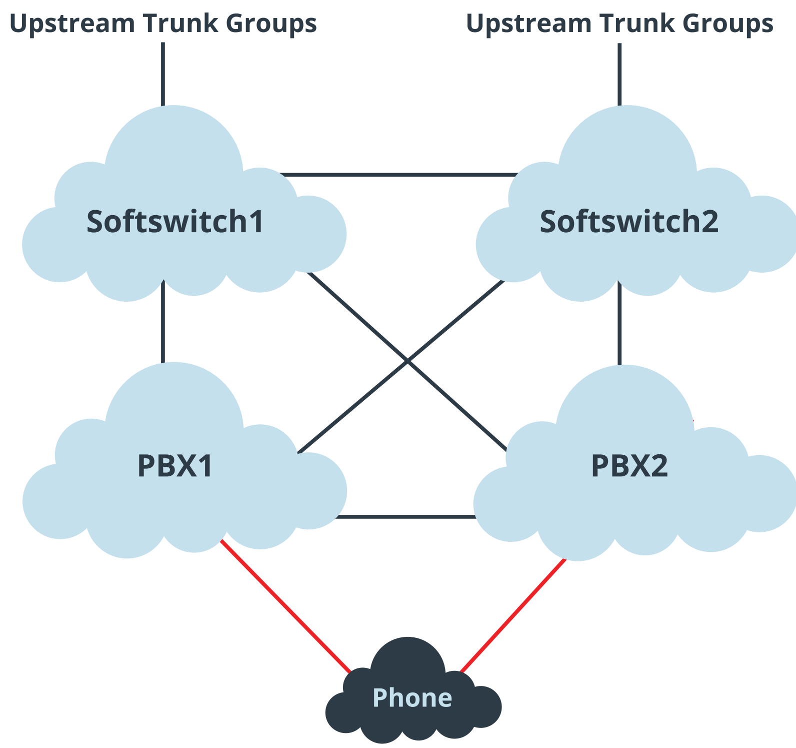
In its simplest implementation, this type of HA clustering requires the use of two upstream softswitches from the service provider and two PBX or B2BUA devices. This simplified configuration looks like the diagram to the right.

In the model, the phone has an identity registered with each PBX. The PBXs synchronize their configurations and data regularly, including call detail reporting (CDR), Voicemail, extension and group settings, and more. The PBXs have a SIP trunk between each other and one to each upstream softswitch.



More Complex Configurations

Using the simple configuration on the next page as a basis, this solution can be scaled out indefinitely by adding additional nodes where needed and extending the mesh. Depending on realistic failure conditions, additional trunking may not need to be 'full mesh' to be effective. For example, an additional PBX would not likely need a trunk to both existing PBXs in order to be effective. It can simply be chained off of PBX2 as the next node in sequence.

This is not the only possible HA configuration for SIP. By introducing additional hardware such as link balancers and load balancers, HA can be achieved at lower layers of the OSI network model. However, the implementation described above is a highly effective solution that is simple to configure and requires no additional hardware or software.

On the next page are some examples of failover scenarios and how they are handled by priority routing on the mesh. If you have any questions about our high availability voice solution, contact our Telephony team at: **Telephony@SierraExperts.com.**

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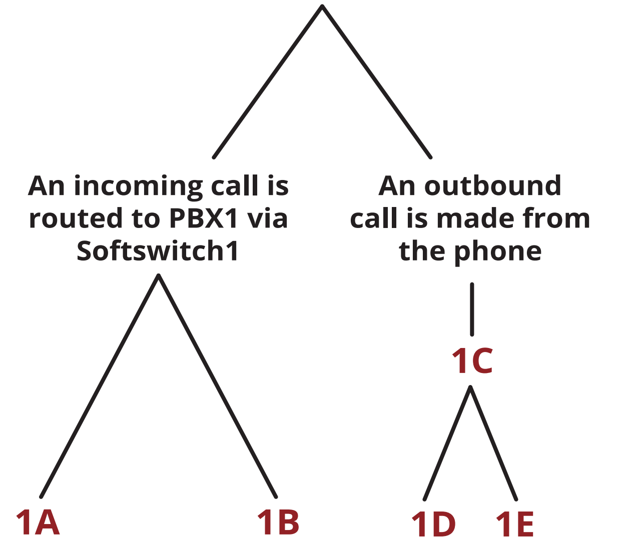
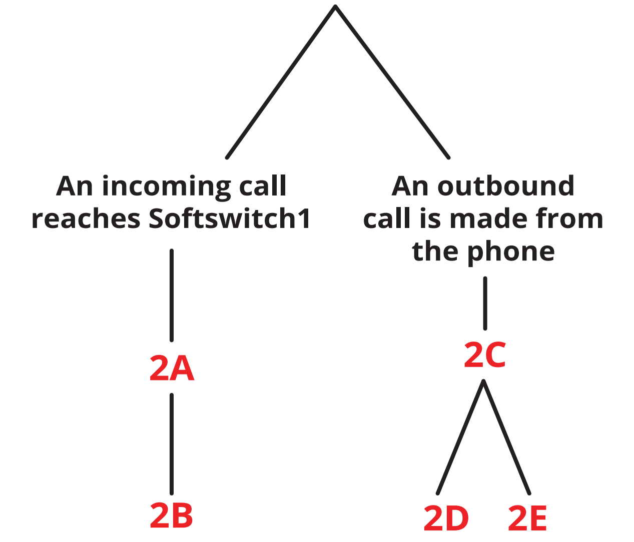
THE PHONE'S REGISTRATION OR CONNECTION TO PBX1 FAILS



PBX1, OR THE TRUNK BETWEEN PBX1 AND SOFTSWITCH1, FAILS



1A
PBX1 sees that the phone is unavailable and attempts to route the calls to PBX2. The call is successfully routed to PBX2, and is sent to the phone.

1B
In the event that PBX2 cannot be reached from PBX1, the call is sent to Softswitch2 instead. Softswitch2 attempts to send it to PBX1 which will fail, then send to PBX2 subsequently to the phone.

1C
The phone, aware that its registration to PBX1 is unavailable, sends the call on its second identity to PBX2.

1D
PBX2 sends the call to Softswitch1, and out to the upstream trunk group to the PSTN2.

1E
In the event that PBX2 cannot reach Softswitch1, the call is sent to Softswitch2 instead, which routes it out to the upstream trunk group to the PSTN.

2A
Softswitch1 fails to route the call via PBX1 and routes it to PBX2.

2B
PBX2 sends the call to the phone.

2C
The phone, aware that its registration to PBX1 is unavailable, sends the call on its second identity to PBX2.

2D
PBX2 sends the call to Softswitch1 and out to the upstream trunk group to the PSTN.

2E
In the event that PBX2 cannot reach Softswitch1, the call is sent to Softswitch2 instead, which routes it out the upstream trunk group to the PSTN.

Going upstream, failures at the switch level are handled the same way. If a softswitch or its trunks become unavailable, inbound and outbound calls are simply rerouted through the second softswitch.

Contact us today to learn more.

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