The Hybrid Approach to VoIP Deployment

Abstract

This note overviews the contemporary approaches to VoIP. It highlights the major pros and cons of “On Premise” and “Cloud” or “Hosted” based deployments - the predominant deployment methods today - and then shows how the benefits of these approaches can be combined in a new Hybrid technology that has been introduced by Bluestone.

Today’s VoIP Deployments

PBX deployment today is premise based or cloud based. Both had their pro's and con's - these are summarized in the Table 1.

- Put the PBX on premise and installation, NAT and firewall are simpler, but a server or WAN outage and your customers will get “dead air” when they call you.
- Cloud based solutions do not have this limitation, but phone installation, prioritization of voice, NAT and firewall configuration are difficult. The majority of installation engineers are NOT sufficiently familiar with the issues of deploying a real time, networked service (such as VoIP) to make such installations, leading to poor quality speech and dissatisfied customers.

Today which ever approach you take you are likely to face major issues.

On Premise		Cloud Based	
Pro	**Con**	**Pro**	**Con**
NAT is not normally needed and firewalling is straightforward.	Large capital expense *(typ $500 to $1500 per phone for systems > 20 phones)*	Initial capital expense is minimized.	As all phone traffic flows over the WAN firewall and NAT issues are usually complex.
Can accommodate legacy PSTN connectivity (ie PRI)	Hard to provide backup, usually non-existent.	Backup is usually available.	Complaints of broken up speech are common *(consequence of improper installation as mentioned above)*
Voice quality issues are rare.	Problems or major expense with multi-site systems.	Multi-site is easily accommodated.	Provider can be very remote from your office. High number of router hops can cause speech quality issues.

Table 1 - A comparison of the the “On Premise” and “Cloud/Hosted” VoIP deployment approaches

The Hybrid Approach - combines the best of both Cloud and On-Premise.

Given that the strengths of "on-prem" are the weaknesses of "cloud" and vice-versa would it not make sense to see if there was a way to combine the two?

To do this we would need to:

- Have a server on premise but make sure there was a backup if it or the WAN failed
- This backup would need to have a mirrored version of the dial plan, track announcements etc of the on site server.

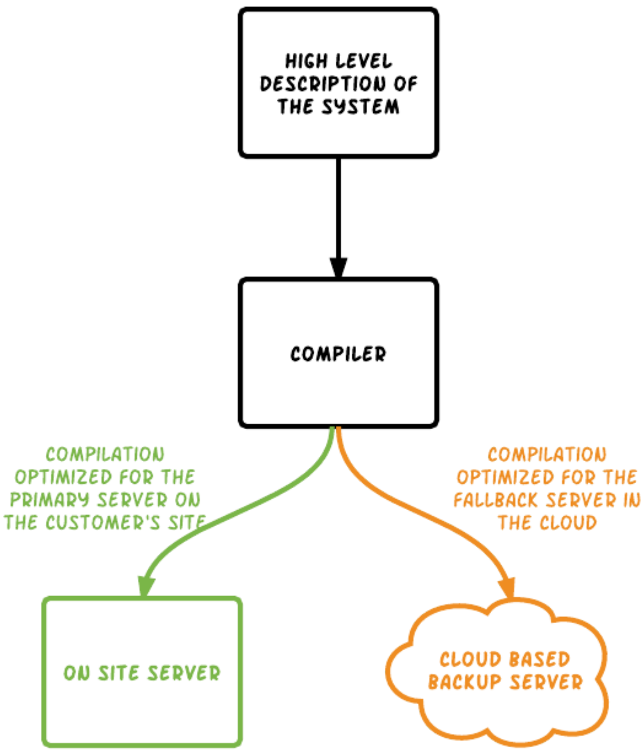

Figure 1 - Bluestone compiler based approach can generate both “on premise” and backup cloud based deployments that always share the same dial plan.

This is in essence what Bluestone has created. To do this we need to have a description of the phone system that could be targeted at two very different targets, the OSS in the environment of the customer's office and in the cloud. These deployments need to have identical dial plans, messaging etc but obviously the one in the cloud is not going to be connected to IP phones as (most likely) the WAN connection to the customer's site is down, so it needs to optimized so that cell or landline phones replace IP phones.

To do all of these things we need a high level description of the phone system that can then be compiled to meet the needs of these two different environments. That is exactly what Bluestone has developed and a simplified flow of the process is shown in Figure 1.

Equipping the on site server with installation smarts solves another problem

We have referred to the complexities of installing voice, now we have a server on site we can the necessary smarts into that server to perform the installation automatically. This has proved in practise to be a HUGE time saver for resellers and IT staff, plus you know the job has been done correctly.

Wrapping it all up

Figure 2 illustrates in practical terms how the hybrid approach works. In normal operation calls from the SIP trunking servers are sent to the on site server (OSS) where they are handled locally. The majority of calls are going to local SIP phones although calls may be forwarded to cell phones and the like as part of "find me follow me" call handling.
If the WAN connection or even the OSS should fail then this is detected by the SIP trunking servers and calls are instead sent to the fallback server in the cloud. This fallback server is know to have an identical dial plan to the primary server and so the customer calling in has no idea that your on site facility is currently down. Calls are sent through to cell phones and the voice mail system. Once the fault has been repaired calls are instantly re-directed back the normal system. There is no manual intervention required.

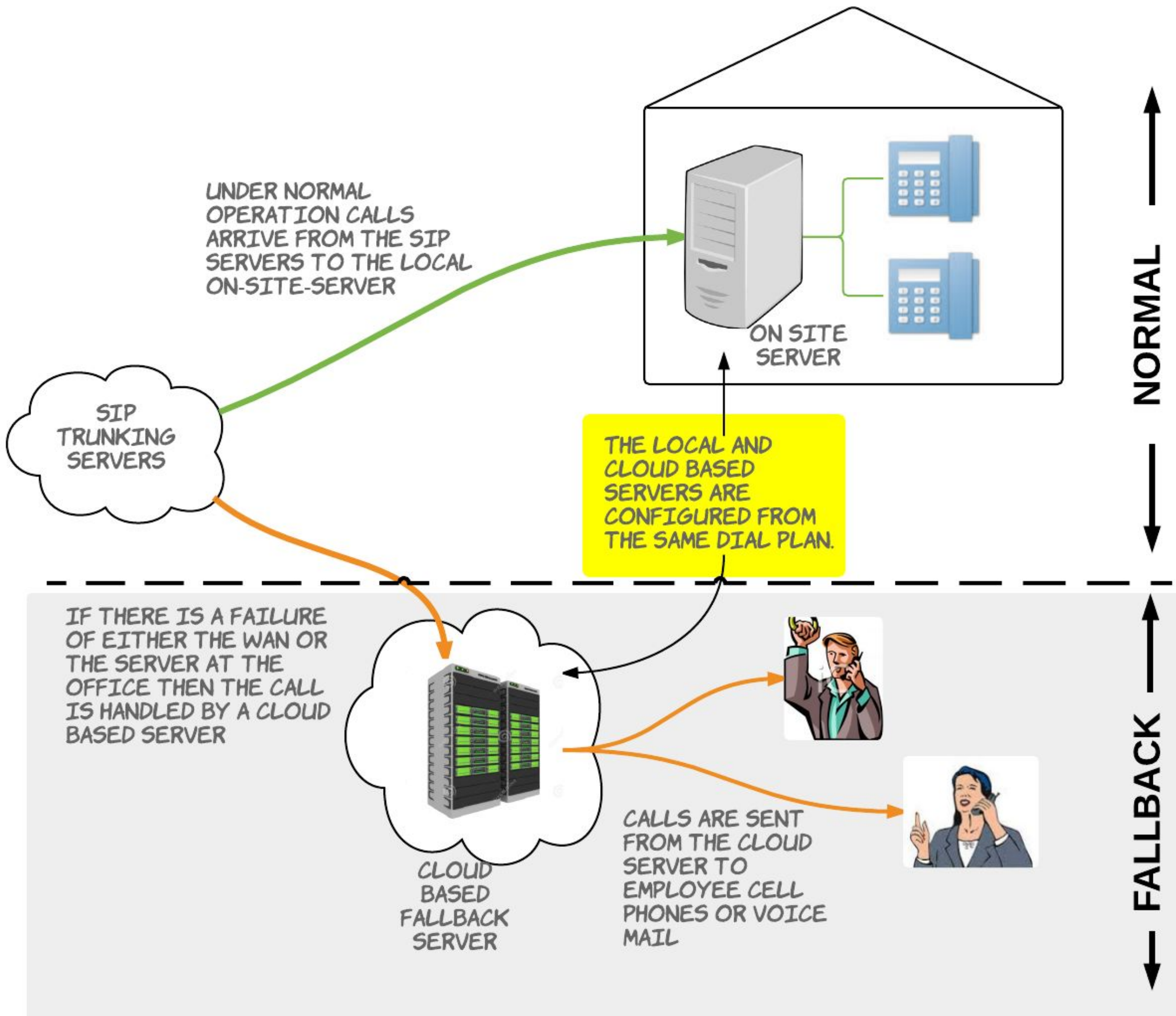

Figure 2 - An illustration of how calls are instantly re-routed should the primary server or it's Internet connection fail

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The On Site Server

This final section overviews the functionality provided by the on site server. These are Intel based units running Linux as the operating system and have the “smart install” and PBX primary server functionality pre-installed. They handle all of the following:

- A broad range of network deployment modes that cover the vast majority of customers needs.
- Network optimization, whereby voice is automatically given priority over data, and QOS is maximised.
- Interface to legacy telecom standards (ir PRI connections) if needed.
- All of the local primary server functionality.