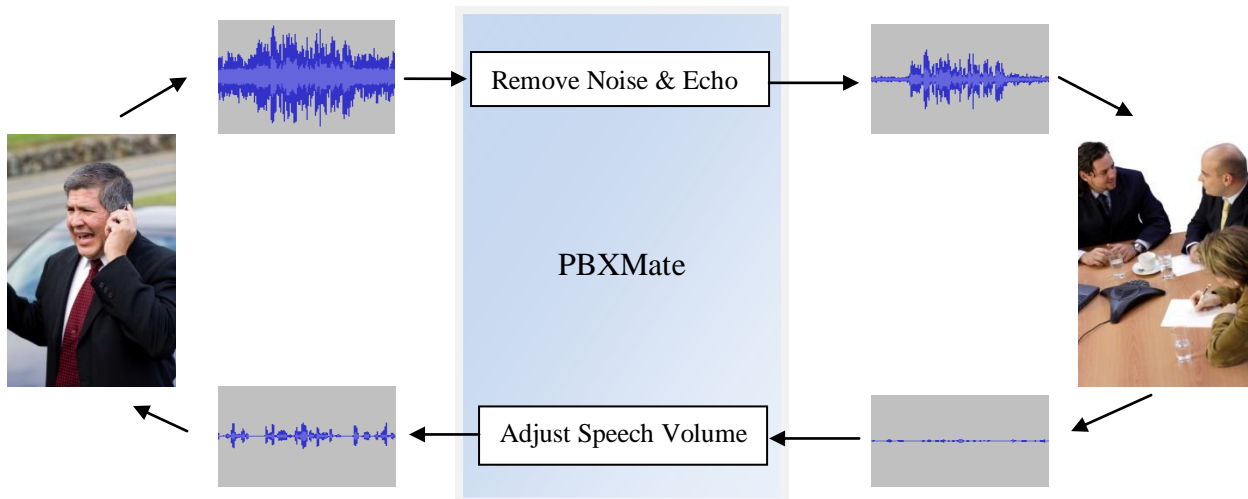


Improve Audio Quality in your VoIP Networks

Does your VoIP network suffer from poor audio quality? Unfortunately, in most cases the answer is YES. Different reasons cause this problem and some of them are due to external factors that you cannot control such as echo and noise originated from the far end. SoliCall's **PBXMate** will allow you to gain control over the audio quality in your network.

What is the PBXMate?

The PBXMate is a software product that improves audio quality for all participants in a call. The PBXMate constantly improves and monitors quality.



Noise and echo are removed and speech level is adjusted to a comfortable level

General			Quality (Extreme)													
Origin	Destination	Duration	Origin -> Destination							Destination -> Origin						
			Noise Level	Max AGC Coef	Echo Level	Jitter (ms)	Packet Loss (%)	Min Delay (ms)	MOS Score	Noise Level	Max AGC Coef	Echo Level	Jitter (ms)	Packet Loss (%)	Min Delay (ms)	MOS Score
3@192.168.0.113	2@192.168.0.145	36	16	100	720	0	0.11	44	4.31	9728	500	1	0	0.15	31	4.29

The PBXMate displays real-time statistics on the quality of the calls

Use Cases

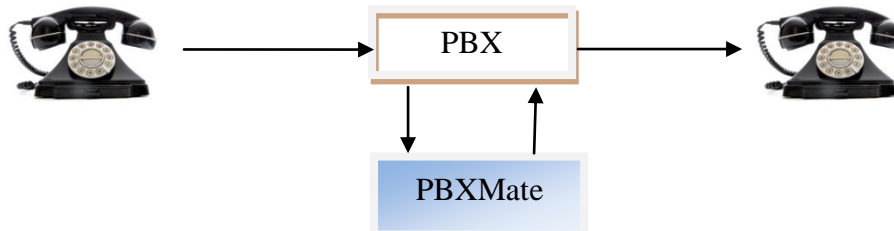
- Remove echo your users hear when making long distance calls or calls over WiFi.
- Shield your customers from the ambient noise in your call center.
- Enhance the audio quality in conference calls with multiple participants.
- Reduce background noise coming from external cellular phones.
- Maintain a comfortable audio volume at conference rooms regardless of the distance between the speakers and the microphone.
- Improve the accuracy of speech recognition engines.
- Monitor calls quality to alert, in real time, on low quality calls.
- Record calls going through the network.

Plug-And-Play Architectures

The PBXMate supports three plug-and-play architectures.

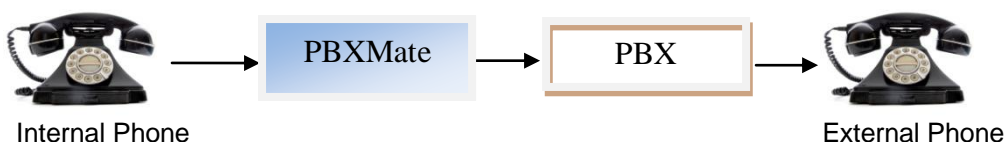
PBXMate as a SIP Trunk (the default architecture)

PBXMate registers as SIP Trunk to the PBX. When the PBX receives a call that needs to be filtered, it routes it to the PBXMate SIP Trunk which, in return, dials back to the PBX. Using the dial-plan in the PBX, the administrator controls which calls are filtered.



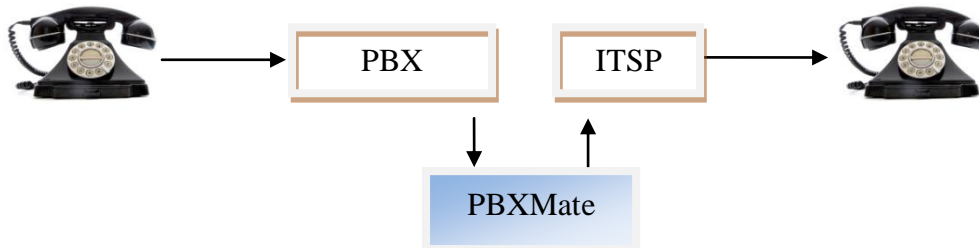
PBXMate as a SIP Proxy

The PBXMate acts as the Sip Proxy for the internal IP Phones. It filters all the calls of these phones. This architecture is **fully transparent** to the PBX and there is no need to do any change in the PBX's dial-plan or in any other network component.



PBXMate as a Router

The PBXMate routes the calls to their destination based on a simple routing plan. This architecture is very similar to the first one (Sip Trunk), but in this case the PBXMate dials to the destination instead of dialing back to the PBX. This architecture can be used when due to licensing issues you do not want to increase the total number of calls in your PBX.



Scalability

The PBXMate can handle hundreds of concurrent calls on a single server. A single PBXMate process splits the load between multiple threads. In addition, multiple instances of the PBXMate can be run on the same server and on multiple servers. The exact number of concurrent calls that can be supported on a single server depends on: the desired configuration (e.g. single side filtering vs. double side filtering), the algorithms that are enabled (e.g. noise reduction, AEC, basic AEC, AGC) and the hardware specifications.

24/7 Uptime

The PBXMate is a robust product which contains a built it recovery & instance mechanism to support 24/7 up time. In addition, in case the PBXMate is being shut down, the PBX will automatically divert all incoming calls directly to their destination without any downtime to the VoIP network.

Supported Networks

As the PBXMate uses the SIP and RTP standards, it works with all VoIP networks that can support these standards. There is also an option to integrate the PBXMate with H.323 network – please consult the user manual for more details.

Supported Platforms

The PBXMate can run on both Linux and Windows. It has both 32bit version and 64bit version. It can be configured to run as a service (Windows) or daemon (Linux). The PBXMate has a flexible port mapping that allows it to run on the same machine that is running the PBX.

Additional Technical Facts

- Supports both web based graphical interface and command line interface.
- Equipped with algorithms to overcome packet loss & jitter.
- Full support for video in two modes: bypass or multiplexed on existing ports.
- Can record all statistics to a file for offline analysis.
- Can fully impersonate the originating SIP phone.
- Adds a minimal delay, about 16ms, to the call.
- Filtering mode can be changed during the call via DTMF controls.
- Allows registering voice sample for additional enhancement in audio quality.
- Can be activated in monitoring mode – i.e. no filtering will be made to the audio.
- Supports different configuration per caller id.
- Allows denial of service for specific caller ids. The PBXMate will return a 503 SIP message in these cases.
- The basic version supports G.711 & G.722. For additional codecs, please contact support.
- Can be configured as a hosted solution.
- Includes a detailed user manual and handful configuration samples & tips.

Customer's Experience

The PBXMate is successfully improving audio quality in many production environments. It is being used by many companies that were unwilling to compromise on quality, ready to invest in order to increase customer satisfaction and therefore add to their competitive edge. Among our customers are call centers, mobile operators and unified-communication platform providers. Case studies and reference accounts are available upon request.