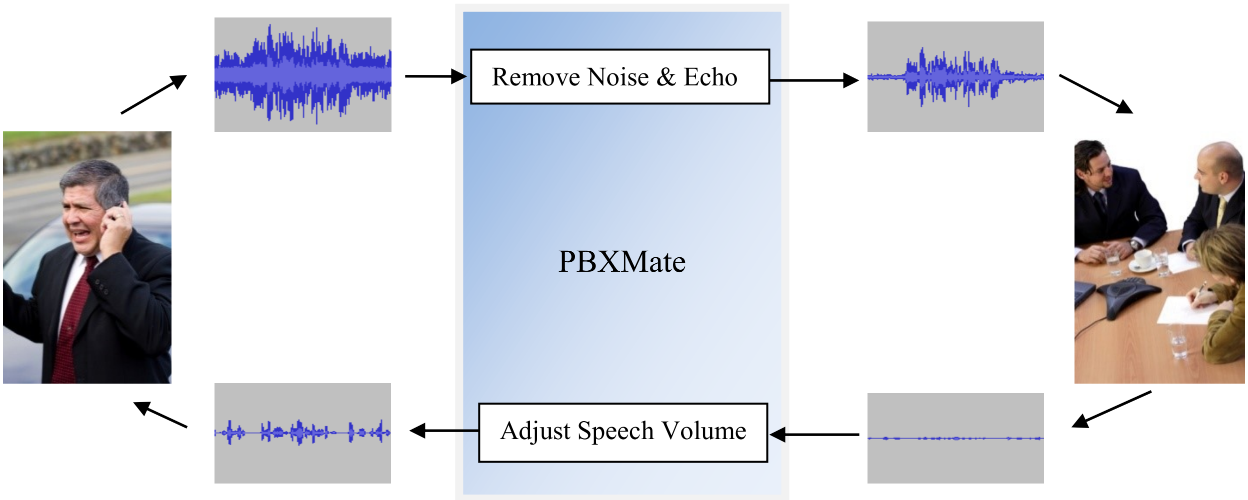
Improve Audio Quality in your VoIP Networks

Does your network suffer from poor audio quality? Unfortunately, in most cases the answer is positive. 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** allows you to gain control over the audio quality in your network.

What is PBXMate?

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



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

General			Quality (Average)															
			Origin -> Destination								Destination -> Origin							
Origin	Destination	Duration	Noise Level	AGC Coef	Echo Level	Jitter (ms)	Packet Loss (%)	Min Delay (ms)	RNR	MOS Score	Noise Level	AGC Coef	Echo Level	Jitter (ms)	Packet Loss (%)	Min Delay (ms)	RNR	MOS Score
2@192.168.0.137	4@192.168.0.137	169	34	100	0	0	0.00	4		4.70	29	100	0	0	0.00	22	0	4.41
3@192.168.0.137	5@192.168.0.137	73	3215	100	0	0	0.00	4		4.70	10	100	0	0	0.00	2		4.41

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

Use Cases

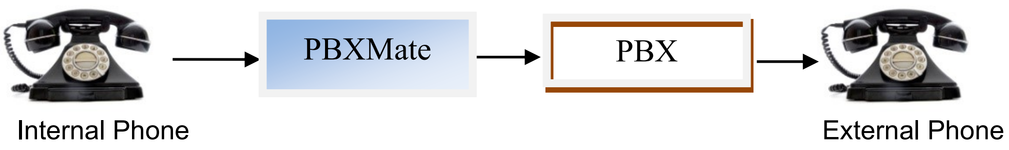
- Shield your customers from the ambient noise in your call center.
- Enhance the audio quality in conference calls with multiple participants.
- Remove echo your users hear when making long distance calls or calls over Wi-Fi.
- Reduce background noise coming from external cellular phones.
- Maintain a comfortable audio volume.
- Monitor calls quality.
- Improve the accuracy of speech recognition engines.
- Record calls.

Plug-And-Play Architectures

PBXMate supports three plug-and-play architectures.

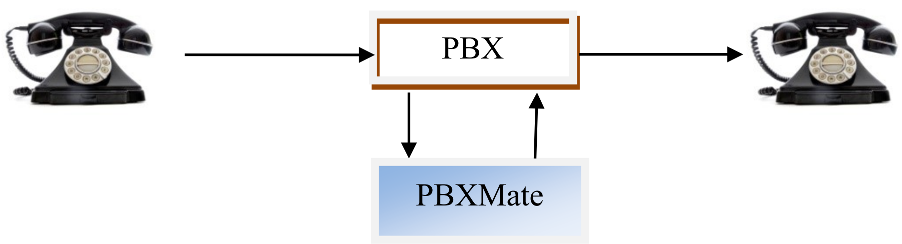
PBXMate as a SIP Proxy

PBXMate acts as the Sip Proxy for the internal IP Phones. It filters all the calls of these phones. This architecture is **fully transparent** and there is no need to do any change in network components.



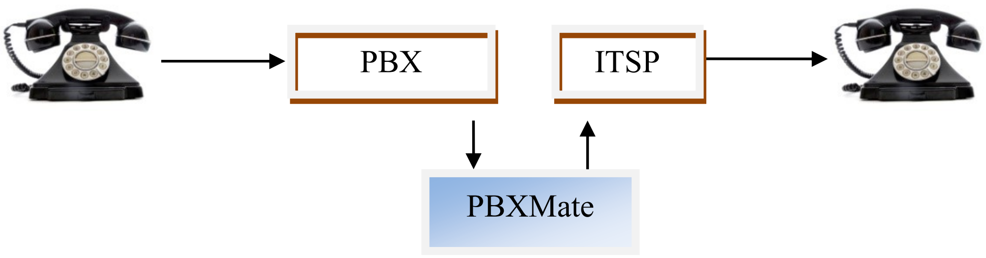
PBXMate as a SIP Trunk

PBXMate registers as SIP Trunk to the PBX. When the PBX receives a call that needs to be filtered, it routes it to 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 Router

PBXMate routes the calls to their destination based on a simple routing plan. This architecture is very similar to the previous 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

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) and the hardware specifications.

24/7 Uptime

PBXMate is a robust product which contains a built in recovery mechanism to support 24/7 up time. In addition, in case PBXMate is being shut down for maintenance, the PBX will automatically divert all incoming calls directly to their destination without any downtime (in Sip Trunk architecture).

Supported Networks

PBXMate uses the SIP/RTP standards and works with all VoIP networks that support these standards.

Supported Platforms

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

Additional Technical Facts

- Includes multiple noise reduction algorithms.
- Includes advanced server-side echo cancellation.
- Includes automatic gain control.
- 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 be controlled using DTMF.
- Can be activated in monitoring mode – i.e. no filtering will be made to the audio.
- Allows denial of service for specific caller ids.
- The standard version supports G.711 & G.722. For additional codecs, please contact us.
- Includes a detailed user manual, configuration samples and tips.

Customer's Experience

PBXMate is successfully improving audio quality in numerous production environments. It is being used by many companies that are unwilling to compromise on quality. Among our customers are call centers, conference bridge providers, operators and unified-communication platform providers. Case studies are available upon request.