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Echo Cancellation in Voice Communication Systems

[1]Forhad Monjur Hasan, [2]Onell Allan Chakawuya and [3]Mostafa Mohammad Razaul

[1-3]Department of Electronics and Information Engineering, China West Normal University, China

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Corresponding Author: Forhad Monjur Hasan

Abstract

Echo cancellation is an essential part of present-day voice communication systems, and it has gained relevance against the background of the worldwide popularity of VoIP, teleconferencing, and hand-free, which are necessary to operate. Echo A delayed repetition of the voice of the speaker, known as such, may significantly worsen the quality of the hearing and hinder the efficiency of the communication process. The paper at hand thus also represents an in-depth analysis of the concepts and problems associated with echo cancellation. It deals with how echo is varied and what causes it, how adaptive filtering algorithms are designed and work and how their operation is managed, how residual echo can be controlled and double-talk cases can be handled, the practicalities of implementation, performance measures, to the newer adaptation of machine learning. The entire purpose will be to give an overall survey of present methodologies and to outline future avenues of development.

Keywords: Adaptive Filtering, Double-Talk Detection, Echo Cancellation, Kalman Filter, Machine Learning, Voice Communication

1. Introduction

Echo is one of the perennial failures in voice communications system taking the form of an echo effect that delays the speaker voice. And echo effect may interfere with the flow of a conversation and Exploit speech intelligibility [1]. The need to improve the quality of communication to the high fidelity levels has increased as applications like the voice over IP, teleconferencing, mobile communication and smart voice interfaces are increasing it hence the effective echo cancellation has become paramount. The impedance asymmetries found in traditional circuit-switched infrastructure (such as PSTN) will cause the phenomenon of the hybrid echo, whereas the network induced delays and jitter in a packet-switched infrastructure (such as VoIP) will make the echoes even more audible. To counter these effect echo cancelation is employed to essentially remove the echo signal by capturing and subtracting it using signal processing [2]. This paper presents a component study of the nature of echo, its physical and perceptual properties and gives an insight into the mathematical models adopted to estimate the echo path. It also looks into adaptive filtering methods including LMS and RLS and Kalman filters, and more superior methods of handling both double-talk and residual echo. The difficulties of implementation which concern computational limitations, DSP architecture, and real-time operation are also discussed. Lastly, the article covers evaluation criteria, application cases, and new research directions that give an in-depth background to development and perfection of echo cancellation systems.

2. Fundamentals of Echo

Echo in voice communication Echo is some of the transmitted audio signal being reflected back to the source. This is normally delayed. This is particularly disadvantageous under bi-directional configuration of communication where the signal passes, and at the same time, it is being received [3]. The major causes of echo are impedance mismatch, coupling and signal processing delays. The strength and perceptibility of the echo is explained by the delay time, which occurs, as well as the strength of the signal which is reflected and the sensitivity of the human hearing. Echo divides into three main categories namely acoustic echo, hybrid (line) and network echo. In speakerphones and conferencing systems, acoustic echo tends to be common where the output of the loudspeaker is being picked up by a microphone that is close by. Such echo is determined by sounding in a room, the location of speakers and microphones and the quality of the hardware. Hybrid echo occurs, however, frequently in the old analog telephone systems. It is caused by the impedance mismatch between the two-wire part of the telephone network and the 4 wire part, so that some of the signal is reflected at the point of connection to the hybrid [4]. This type of echo is mostly evident on long-distance calls as well as on systems that do not have the requisite echo suppression infrastructure. Echo in networks A network echo may occur in a digital network such as used in voice over IP (VoIP) that reintroduces previously sent audio signal due to delay, jitter, or codec processing across the system creating an echo.

Effects of echo on the quality of communication are perceptual and technical. Perceptually, echo may be very distracting to the users such that they end up with impaired intelligibility, augmented cognitive load and overall appalling communication process. Echo technically adds unnecessary signal power that makes signal processing activities difficult, e.g. speech recognition and audio enhancement [5]. Echo is especially troublesome when delay of more than 25 milliseconds occurs in both directions so that the detection and suppression becomes an important part of the design of voice communication systems. Effective echo cancellation solutions should, therefore, detect, suppress the echo effectively, as well as working in varying conditions such as double-talk conditions and non-linear industrial paths.

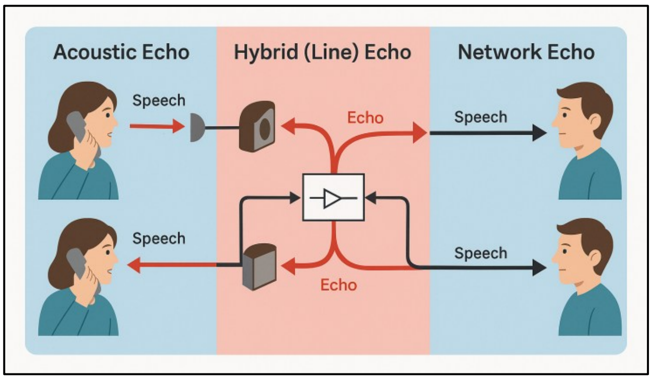

Figure 1: The fundamentals of echo

3. Echo Path Modeling

Echo path modeling is central to successful echo cancellation algorithms, since it entails representing the path taken by the echo signal, as traversing the architecture right up to where it is re-entering the communication system. The chain of the echo usually contains the acoustic environment, the electro acoustic transducer (speakers and microphones), an analog front-end, and the digital processing. Knowledge of this route is essential as it defines straight away the attributes of the echo and the formation of factual mitigation schemes.

Physically, the echo path does not only consist of the straight occurrence of the loudspeaker to the microphone but a variety of reflected channels caused by physical structures such as walls, ceilings, and objects in the room. These thoughts are reflected towards the impulse response of the system which can be long and complicated [6]. The echo path is naturally a time-varying property since it is subjected to moving users, evolving environmental conditions, and even relocations of the devices that contributes to the complexity of its modeling.

The echo path can be mathematically approximated by a finite impulse response (FIR) filter as an operation where the impulse response is often assumed to convolve the original far-end signal to become the echo [7]. This Linear convolution model provides a theoretical basis on adaptive filter-based adaptive cancellation where an estimate of the unknown impulse response is required in real time. Let:

x(n): Far-end signal
h(k): Impulse response (echo path)
y(n): Echo signal
N: Filter length

$$y(n) = \sum_{K-0}^{N-1} h(k).x(n-k)$$

This equation demonstrates the echo modeling through the use of convolution of the signal at the far-end and the impulse response of the system.The linear model can be however insufficient because of the non-linearity introduced by the hardware component e.g. microphones, speakers and amplifiers.

Adaptive algorithms have to update filter coefficients continually to follow changes in the impulse response to adapt to the constantly changing nature of the echo path. It requires the system to be resilient, sensitive and compute-efficient. Such systems can be designed with trade-offs among filter length (it must be long enough to pick up all the echo path delay), convergence speed and numerical soundness [8].

A precise echo path model gives this system the capability of anticipating and removing the echo before it is received thus increases the quality of the communication. Artifacts and/or residual echoes may be heard due to improperly modeled echo paths and this decreases the overall usefulness of the cancellation algorithm [9]. Hence, knowledge of echo paths structure and behavior is needed when designing algorithms, as well as systems.

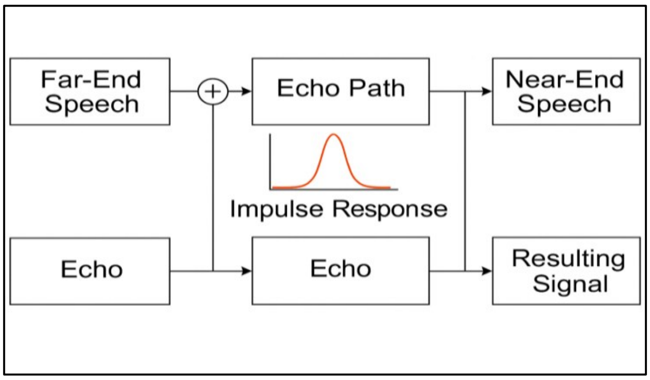

Fig 2: Block diagram of echo path modeling

4. Adaptive Filtering Techniques

The modern day echo cancellation systems work around adaptive filtering techniques. These methods allow dynamic estimation and removing of the echo signals by constantly revising filter parameter according to inbound information [10]. There is an underlying principle of modeling the unascertained echo path and updating the filtering coefficients in real time so that the predicted echo error and the actual echo awesome movement are reduced. Good adaptive filters have to converge fast, stay steady under conditions of double-talk and perform within computational limits.

A well known adaptive filter, the Least-Mean-Squares (LMS) algorithm differs in that it is simple and has a relatively low computational cost. LMS adjusts its filter coefficients as they minimize the mean square difference between the requested and observed signal. Although LMS is easy to implement because of the advantages, it has a slow convergence rate and sensitivity to variations of signal power that may slow performance in dynamic conditions [11].

To limit this the Normalized Least-Mean-Squares (NLMS) algorithm adds an additional normalization coefficient over the power of the input signal and thus stabilizes the update and increases convergence [12]. NLMS performs especially well in system with time dependence signal amplitude that is large.

Recursive Least Squares (RLS) algorithm has far earlier convergence than LMS and NLMS and RLS minimizes the weighted least square error of previous signal samples. The computing resources needed by RLS, however, is more than that required by the three other protocols and this would be too hectic to implement in a system that has limited processing resources or has less rigorous requirements. It is especially adaptive in scenarios in which the path of the echo varies fast [13].

And still in more complicated and dynamic environments more sophisticated algorithms come in play like the Affine Projection Algorithm (APA) and the Kalman Filter. APA generalizes NLMS, employing more than one past input vector in performing the adaptation such that it has better convergence characteristics. Kalman Filter gives the best results when used in linear dynamic systems because it explicitly models the statistical behavior both of the system and noise involved which is why it is the best when the accurateness of the model and the outcome of this model are of the greatest concern.

The choice of particular adaptive filtering method Chris Ho (2001) has to rely on several factors such as the tolerance on system latency, the amount of available computational resources, desired fast convergence rates, and assumptions on the variability of the echo path.

Table 1 summarizes in a comparative fashion five Adaptive filtering algorithms commonly used in echo cancellation to help in the selection of an appropriate adaptive filtering method. The methods are compared according to their computational complexity, rate of convergence, stability and common examples of applications. Being computationally efficient, the LMS algorithm takes a long time to converge and it also is most appropriate in systems with stable echob paths. Compared to LMS, NLMS is an improved version which takes consideration of the power of the input signal leading to a quicker and stable convergence. The disadvantage is that RLS is slow to adapt and precise but with a higher computational load and thus suited to dynamic settings. APA is a mid-way between complexity and performance, since it uses several input vectors when adapting. The Kalman filter, the most computationally complex one, offers the best results when it comes to linear dynamic systems when specific accuracy in the estimation of states is required. This analogy assists designers to trade off performance versus hardware limitations and real- time processing need in their actual applications.

In practical usage, there is usually a trade-off between algorithmic complexity and echo cancellation.

Table 1: Adaptive Filtering Algorithms: Trade-offs in Complexity, Speed, and Stability

Algorithm	Complexity	Convergence Speed	Stability	Suitability
LMS	Low	Slow	Moderate	Simple systems
NLMS	Low	Moderate	Improved	Varying signal amplitudes
RLS	High	Fast	High	Rapidly changing conditions
APA	Moderate	Faster than NLMS	Good	Multi-vector environments
Kalman Filter	Very High	Optimal	Very High	Precise dynamic systems

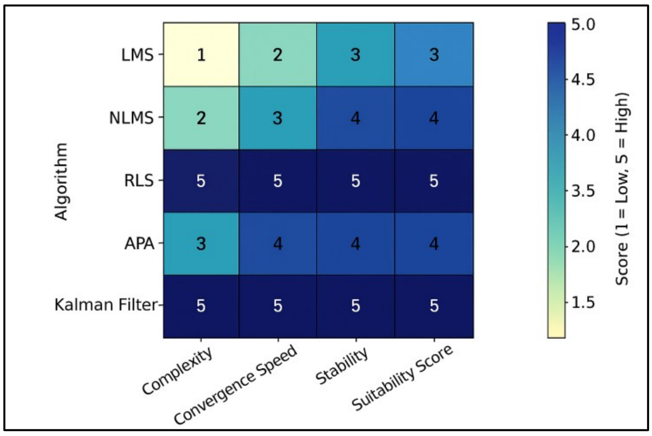

Figure 3: Presents a heat map comparison of key adaptive filtering algorithms

A heat map comparison of major adaptive filtering algorithms is given in Figure 3. It gives the visual depiction of the trade-offs on the complexity, speed of convergence, stability and allows one to select appropriate techniques to complex echo cancellation problems.

5. Double-Talk Detection and Control

Adaptive echo cancellation systems component Double-talk detection and control is important to the robustness and stability of the filter adaptation process when conversational overlap occurs. The situation in which both the far end user and the near end user talk at the same time is known as the double-talk [14]. Under these conditions, when the adaptive filter keeps updating its coefficients, there are chances of misinterpreting the near-end speech signal as being part of the echo signal. This may result in divergence or distortion of the echo canceller which induces a great degradation to

the system performance.
To ensure the integrity of adaptive filtering in conditions of a double-talk, proper identification of the occurrence of double-talk and accordingly pause or modify the filter adaption process is very necessary. The energy levels of a near-end and far-end signal may be compared as one of the simplest though effective means. Substantial increase in the near end signal energy, compared to expected echo is a probable sign of double-talk. Cross-correlation analysis is also another common technique, where the microphone signal is correlated to the far-end reference signal to determine the correlation. Double-talk is normally indicated by a low correlation value with large levels of near-end activity [15].
State-of-art detection mechanisms apply statistical features and machine learning classifiers. Using these models, temporal and spectral characteristics are used to differentiate between near-end and echo speech more successfully [16]. To give an example, we can train Gaussian Mixture Models (GMMs) or Support Vector Machines (SVMs) with labeled conversational data and this can make the detection show great reliability particularly in the noisy or unpredictable settings.
When the control mechanism detects it, then it has to freeze the adaptation process or to move into a strong-update mode with limited impact on the near-end signal. Other contemporary systems use a softer adaptation strategy, and slow down but do not stop the adaptation rate, so filter responsiveness is preserved without sacrificing stability.
The success of echo cancellers in noise reduction is not the only advantage of good double-talk detection and control but also preconditioning the enhancement of user experience since the conversation flow and sound preservation between two or more speakers remain natural. The realization of speech overlap is essential in practical applications like VoIP, cell phones, and voice-controlled interface where speeches are often interfered with and it is inevitable.

6. Non-Linear Processing & Post-Filtering

Regardless of adaptive filtering efficiency, a certain amount of residual echo usually can still be found because of such reasons as improper filtering section, non-linearities of a hardware component, or the inadequate filter length. Such residual echo may be a disruptor on the perceptual level and will have to be even more squelched down by non-linear processing (NLP) and/or post-filtering. These further steps in the processing are important in making high quality, echo-less communication.
Non-linear processing is whenever signal suppression is employed beyond the adaptive filter. These methods are aimed at reducing the residual echo signal (that cannot be removed by the adaptive filter). Spectral subtraction is a common strategy to apply, where the estimated power of the residual echo is removed (by subtracting) it out of the overall signal spectrum by reducing its audible effect [17]. The other method that is widely applied is the use of suppression masks or the echo suppression gains over the selective attenuators of time-frequency components of the perceived signal to possibly have echo.
Among the drawbacks of non-linear processing is that it becomes hard to differentiate between the genuine speech and the remaining echo of the speech, particularly in conditionss involving a double-talk. Mistaken suppression can mean the discard or alteration of the wanted near-end speech. In order to improve this, current systems use echo presence probability estimator or a machine learning model that would provide better decisions and less distortion of speech.
Besides the act of suppression, post-filtering also includes generating comfort noise. In instances where echo is totally eliminated in silent moments, the end-experience is most likely to sound artificial or rudely silent to individuals. Synthesized background noise generation (CNG) is an auditory tactic that allows accommodating constant auditory surroundings by balancing out background noise to enhance the natural talk [18].
Useable non-linear processing and subsequent filtering is a considerable improvement of communication systems in perceptual terms. Such techniques are essential when designing hands-free devices, conferencing systems and smart speakers, where the echo path may be many-path and variable, sometimes changing periodically. The combination of these methods with adaptive filtering will guarantee full-fledged echo reduction and uninterrupted interaction with the program.

7. Implementation Considerations

To have an echo cancellation in a practical communication system, expert care should be taken on the compute efficiency, hardware limitation, and flexibility of integration. Implementations of the echo cancellation algorithm either in hardware or in software should strike an equilibrium between performance, latency and resource requirements.
Computational complexity is one of its main considerations. Post-processing algorithms and adaptive filtering, most notably such advanced filters as RLS and Kalman filters, can cause an excessive processing burden. Processing latency has to be minimised in second-by-second systems like VoIP or cell phone telephony in order to make conversation fluid. Thus, executions tend to use optimized algorithms which cut the number of arithmetic operations without affecting the performance. Such methods as block-based filtering and fixed-point arithmetic are often used to improve efficiency [19].
Implementation selection as a fixed-point or a floating-point is another factor. The need of lower power consumption, fewer memory requirements and arithmetic operations allows greater use of fixed-point digital signal processing (DSP) in embedded systems. Fixed-point arithmetic is however subject to numerical instability and needs precarious methods of scaling and rounding. Floating-point DSP, however, is more precise and has a broader dynamic range, and is thus appropriate to applications requiring customization, prototyping and high-end applications where processing power is not as limited.
Performance and the flexibility of the system are also a factor in the choice presented on whether to do the echo cancellation in software or hardware. Application-specific integrated circuits (ASIC) or field-programmable gate arrays (FPGA) implementations, provide lower latency and increased throughput. These are most suitable in limited resource or real time applications like automobile or embedded telephony systems. On the contrary, the soft

phones, mobile applications and cloud based communications systems favor software-based solutions because of the ease of deployment, scalable capability, and upgradeability.

Furthermore, the integration with the current communication stacks is vital, and optimization on this per platform is essential. As an example, power and latency constraints may be quite severe in mobile operating systems, leading to lightweight implementations which must be able to inter-operate with other processes within a system. Equally, the smart speakers and the conferencing systems might need the echo cancellers to connect them with the beam forming and voice activity detection modules which would create extra complexity in the process.

To conclude, the key to a successful deployment of echo cancellation algorithms would be to balance the trade-offs of the computational demand, accuracy, latency and adaptability. Optimization of the design towards the certain platform and application assures an efficient operation with conservation of user experience and system-efficiency.

8. Performance Metrics & Evaluation

Testing the performance of echo cancellation systems is a mixture of objective, perceptual and subjective tests. They are crucial evaluation criteria that are needed to compare the performance of algorithms, optimize a system, and make the end-users satisfied.

One of the most popular quantitative measures called Echo Return Loss Enhancement (ERLE) measures the amount of signal power reduction caused by the echo canceller. It is measured in dB and is defined as the ratio of the power of the estimated echo signal (platform) prior to cancellation and the power following cancellation. The higher the value of ERLE the more the echo suppression. Nevertheless, the inconvenient pronunciation or pleasure of words cannot be measured utilizing ERLE exclusively [20].

In response to this, metrics that are used to perceptually evaluate the quality of the voice are used including perceptual evaluation of speech quality (PESQ) and perceptual objective listening quality assessment (POLQA). PESQ contrasts original and processed audio-signals depending on a psychoacoustic model approximating perceived quality. Successors to PESQ are the POLQA, specified by the ITU-T as successor to PESQ and occupying a similar niche (the standard is based on wideband and super-wideband signals and are more sensitive to codec artifacts and packet loss). These are metrics which correlate easily with the human perception and are utilized widely in voice over IP (VoIP), cellular telephony, and conference analysis.

Other than the objective measurements, subjective tests of listening cannot be dispensed with. With controlled conditions the human listener measures the quality of speech, its clarity, and the existence of braided echoes. Such tests can give information on artifacts and subtleties that would be missed by automated metrics. It is often employed using standardized methodologies, including the Mean Opinion Score (MOS) test; in the test, the subject ranks audio quality on a fixed scale.

The other additional figures are the latency that influences the natural flow of conversation and signal-to-echo ratio (SER), a measurement of desired speech clarity compared to echo. Besides, it tests double-talk robustness in order to ascertain that the echo suppressions are consistent and precise when both parties are simultaneously in action. A good evaluation scheme would normally provide a combination of several metrics to get an overall picture regarding the performance of the systems. With the help of correlating objective measurement and perceptual and subjective outcomes, developers can optimise algorithms to suit application requirements, be it low-latency mobile communication or high-fidelity conferencing systems.

9. Applications & Case Studies

There are many different platforms related to the use of echo cancellation technologies and each of them needs special attention and specific requirements. The performance of these systems has a lot of influence on the user experience in the consumer and enterprise setting. This segment goes into the major areas of application and practical instances where echo cancellation is highly trendy.

A. Teleconferencing and Hands-Free Telephony: The teleconferencing system has many participants that talk through the use of open microphones and loudspeakers which favor an acoustic echo situation. Echo cancellation is important in preserving normal flow of the conversation by avoiding echo feedback loop which may lead to interference. The higher-order types of echo canceller are built into speakerphones, conference bridges and unified communication systems to achieve clarity in both multi-party conference situations and two-person talking in echo-prone situations. Another important application is automotive, where there are small hardware requirements, high levels of engine noise and varying acoustic conditions, which call on echo cancellation to be highly robust and operate with minimal latency over a range of high acoustic levels to make drivers safe and intelligible.

B. VoIP and Mobile Communications: Voice features and mobile phones- VOI-P services and mobile phones are applications that largely depend on software based echo cancellation. Such systems encounter a problem of jitter, unreliable delays, and codec artifacts which affect echo performance. Mobile operating systems incorporate scene-specific echo cancellers on the system side, rather than the application side, and include voice calls, video conferencing, and virtual assistants as examples of possible use cases. Echo cancellers have to be extremely flexible, bandwidth-conscious, and computationally light so as to be easily customizable as well as effective processing in real-time considering the mobility of users and the fact that the environment can change.

C. Voice Controlled smart speakers and assistants: Smart speakers and voice command assistants such as Amazon Alexa or Google assistants, or Apple Siri all employ far-field microphones to identify speech commands even when premissed with audio playback. This situation poses a very large levels of acoustic echo which necessitates high level of echo cancellations built into the beam forming, reduction of noise as well as wake-word detection. When work with a low signal-to-noise ratio (SNR) the echo canceller should be able to separate the voice of the user and the signal being played in order to have a smooth communications process. The echo path is very dynamic because these devices are operating in various acoustic places (e.g., kitchens, living

rooms). That is why constant adaptation is needed.
These practical cases indicate that it is impossible to avoid implementing echo canceling systems based on unique device limitation, acoustical conditions, and user anticipations. An advanced signal processing and machine learning incorporated in current echo cancellers has greatly improved their flexibility in various platforms and their functionality.

10. Challenges & Future Directions

A lot of strides have been covered in the area of technologies used in echo cancellation but there are still a number of limitations especially in new applications and newer communication scenarios. The next generation of advances should lead to an even higher level of complexity, user requirements in terms of smooth interactions and restrictions created by new-generation platforms.
Echo Cancellation in Full-Duplex Systems: To make a full-duplex transmission work the transmitter and the receiver need the ability to keep on with speaking and receiving at the same time without any delay or noise being noticed too much. This kind of setup puts a strict constraint on echo cancellation systems in that such systems must of necessity be stable and perform in spite of the likelihood of incessant double-talk which in traditional methods with adaptation algorithms has proven to be a problematic trait. This has led to the mainstay of latency and the perceptual metrics to be stringent as the high-degree of versatility in real-time echo cancellers is limited in research and the practical application of the same in the current communication study sphere.
Machine-Learning Approaches: Deep Learning and Machine Learning are promising directions of enhancing the performance of echo cancellation, especially in high-stakes acoustic environments. Neural Networks can identify non-linear attributes of acoustic echo paths, and exhibit robust generalization over a range of environment and apparatus. The architectures for end-to-end structures have been proposed to establish the functions of echo suppression, noise reduction and speech enhancement into a single structure. However, heavy computational overhead, low interpretability and the need to require large labelled datasets are presently limiting their practical use. The ongoing research is aimed to reduce the size of computation and make them capable of being executed in real-time, and making their performance more robust in heterogeneous operating environments.
Echo Control in Multi-Microphone Arrays: Microphone arrays are becoming more and more prevalent in contemporary smart devices in combination with spatial audio processing, beam forming and noise suppression. Due to the nature of such systems where echo cancellation is an inherent part, there occurs extra complexity because of the existence of multiple echo paths, and spatially varying echo sources. The existing studies are therefore aimed at devising procedures where echo control is combined with directional processing. Techniques such as generalized side lobe cancellation (GSC), adaptive beam forming and spatial post filters have been suggested to meet this requirement. Such techniques have to trade off inter-microphone coherence and multi-path effects in maintaining the integrity of the target speech.
Emerging Codecs and Network Conditions: and communications as a whole are moving to newer codecs, e.g., EVS and Opus, and being deployed within dynamically-challenged networks, including 5G and Wi-Fi 6, forcing an echo canceller to mind newer delay shapes, unpredictable packet-loss patterns, and codec-specific impairments. This makes the echo path estimation more complex and slow down filter convergence. Remedies involve adaptive buffer control, jitter compensation methods and codec-sensitive echo modeling.
With that in mind, next-generation echo canceller architectures are expected to empower an information-driven machine-learning based framework, complemented by traditional signal-processing paradigm, ultimately resulting in a scalable, context-sensitive and robust echo suppression across a broader range of devices, not just mobile devices, low-cost conferencing systems, and augmented-reality/virtual-reality installations, but also autonomously operate communicating agents.

11. Conclusion

Echo cancellation has now evolved to become an essential technology with regard to the comprehensibility of speech in our modern voice systems. Its application can be seen in both the long-established networks of traditional telephony systems, voice channels over the internet (VoIP), mobile products, smart speakers, and two-way conferencing support, all of which need quality echo cancellation. A powerful cocktail of adaptive filtering, non-linear signal processing, and perceptual optimization techniques can overcome the entire range of (linear and residual) echoes in real-time acoustically and dynamically diverse environments.
In the given paper, the phenomenon of an echo will be reviewed thoroughly, key filtering algorithms, LMS, NLMS, RLS, and Kalman filters will be surveyed and problems connected with it, including the difficulties of detecting a double-talk, hardware limitations, and the measurement of performance will be identified. In addition, emergent techniques that leverage machine-learning techniques and multi-microphone space modeling are singled out as areas of future echo-control research.
Coincidence of traditional signal-processing skills with data-driven approaches is likely to produce extremely flexible, latency-minimizing, and user-sensitive solutions to echo cancellation in the near future. This will be crucial in enabling the emerging communication systems, such as the augmented-reality and virtual-reality systems, telemedicine, and immersive remote-collaboration systems. Future work in this area will maintain the natural nature, understandability, and the absence of the obtrusive echoes of voice communication irrespective of the device layout, and network particularities.

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